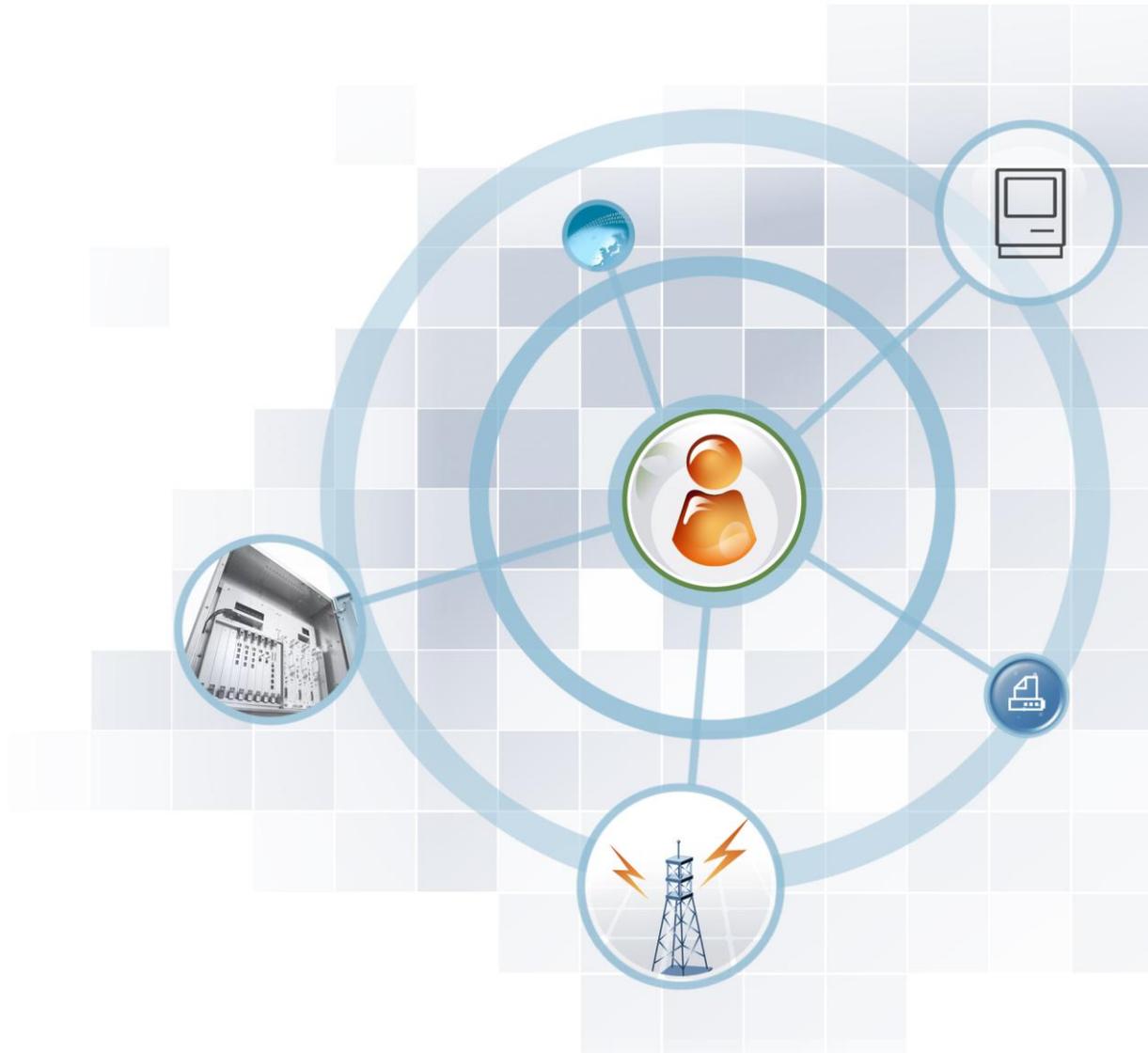


**SCM**

# Quick Installation Guide



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## **TRADEMARKS**

Product names mentioned in this manual may be trademarks and/or registered trademarks of their respective companies.

**This manual should be read and used as a guideline for properly installing and operating the product.**

**All reasonable care has been made to ensure that this document is accurate. If you have any comments on this manual, please contact our documentation centre at the following homepage:**

**Homepage: <http://www.samsungdocs.com>**

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# INTRODUCTION

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## Purpose

This manual describes the quick reference for the system installation.

## Document Content and Organization

This manual consists of the following Chapters.

### Quick Installation Guide

Describes the quick guide for each case during the system installation.

### ABBREVIATION

Describes the acronyms used in this manual.

## Conventions

The following types of paragraphs contain special information that must be carefully read and thoroughly understood. Such information may or may not be enclosed in a rectangular box, separating it from the main text, but is always preceded by an icon and/or a bold title.



### **WARNING**

Provides information or instructions that the reader should follow in order to avoid personal injury or fatality.



### **CAUTION**

Provides information or instructions that the reader should follow in order to avoid a service failure or damage to the system.



CHECK

**CHECKPOINT**

Provides the operator with checkpoints for stable system operation.



NOTE

**NOTE**

Indicates additional information as a reference.

## Console Screen Output

- The lined box with 'Courier New' font will be used to distinguish between the main content and console output screen text.
- '**Bold Courier New**' font will indicate the value entered by the operator on the console screen.

## Revision History

VERSION	DATE OF ISSUE	REMARKS
1.0	10. 2013.	First Version

# SAFETY CONCERNS

The purpose of the Safety Concerns section is to ensure the safety of users and prevent property damage. Please read this document carefully for proper use.

## Symbols



**Caution**

Indication of a general caution



**Restriction**

Indication for prohibiting an action for a product



**Instruction**

Indication for commanding a specifically required action

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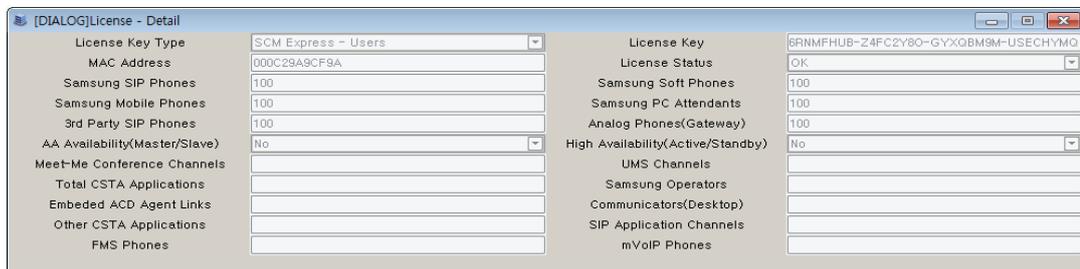
## 1 Interworking local site desktop phone

Depending on the type of phone can be divided into ‘Single Phone User’ and ‘Multi-Extension Phone’. Depending on the type of interworking can be divided into ‘Server Mode’ and ‘PNP Mode’.

### 1.1 Single Phone User (Server Mode)

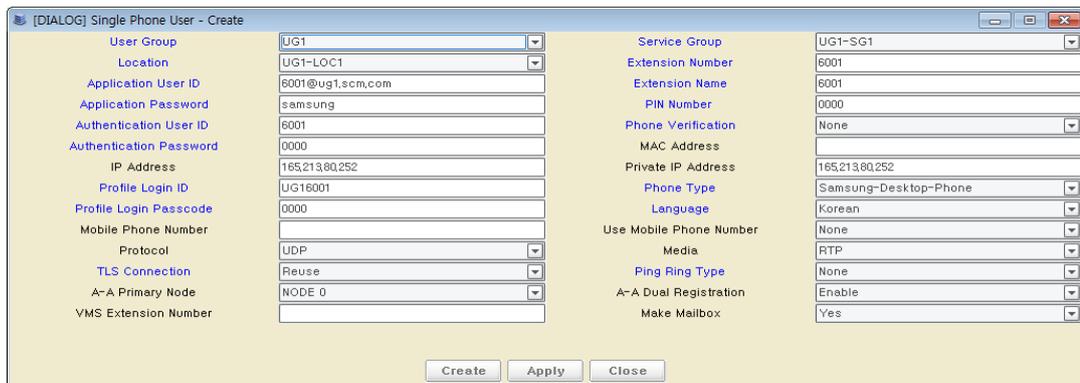
#### License Checking

Check a license count about Samsung SIP Phones.  
**[CONFIGURATION > Miscellaneous > License]**



#### Single Phone User Configuration

Configure the information about Single Phone User.  
**[CONFIGURATION > User > Single Phone User]**



- 1) Select the 'Phone Type' to 'Samsung-Desktop-Phone'.
- 2) 'Extension Number', 'Application User ID' and 'Profile Login ID' are not duplicated with the other user's configuration.
- 3) In Active-Active System case, 'A-A Primary Node' must be set.

### Phone Configuration

Select a 'Configure type' to 'Server' in Easy Install menu of phone.

Enter 'Profile Login ID' in Single Phone User menu into 'Login ID' in Easy Install menu of phone.

Enter 'Profile Login Passcode' in Single Phone User menu into 'Password' in Easy Install menu of phone.

Enter 'SCM IP Address' into 'Config Server' in Easy Install menu of phone.

### Registration Checking

You can check a registration status of phone in next menu.

[CONFIGURATION > Registration Status > Registration Status]

## 1.2 Single Phone User (PNP Mode)

### License Checking

Check a license count about Samsung SIP Phone.

[CONFIGURATION > Miscellaneous > License]

User Group	UG1	Service Group	UG1-SG1
Location	UG1-LOC1	Extension Number	6001
Application User ID	6001@ug1.scm.com	Extension Name	6001
Application Password	samsung	PIN Number	0000
Authentication User ID	6001	Phone Verification	None
Authentication Password	0000	MAC Address	
IP Address	165.213.80.252	Private IP Address	165.213.80.252
Profile Login ID	UG16001	Phone Type	Samsung-Desktop-Phone
Profile Login Passcode	0000	Language	Korean
Mobile Phone Number		Use Mobile Phone Number	None
Protocol	UDP	Media	RTP
TLS Connection	Reuse	Ping Ring Type	None
A-A Primary Node	NODE 0	A-A Dual Registration	Enable
VMS Extension Number		Make Mailbox	Yes

## Single Phone User Configuration

Configure the information about Single Phone User.

[CONFIGURATION > User > Single Phone User]

- 1) Select the 'Phone Type' to 'Samsung-Desktop-Phone'.
- 2) 'Extension Number', 'Application User ID' and 'Profile Login ID' are not duplicated with the other user's configuration.
- 3) In Active-Active System case, 'A-A Primary Node' must be set.
- 4) Select the 'Phone Verification' to 'MACAddress' and enter the MAC address of phone into 'MAC Address'.

## DHCP Server Configuration

Add the next item to DHCP Server.

(Use a SCM IP Address instead of 1.1.1.1)

DHCP Option 43:

[A] code: 43      type: string      data: sec,tftp://1.1.1.1

## Phone Configuration

Select a 'Configure type' to 'PnP' in Easy Install menu of phone.

## Registration Checking

You can check a registration status of phone in next menu.

[CONFIGURATION > Registration Status > Registration Status]

## 1.3 Multi-Extension Phone (Server Mode)

### License Checking

Check a license count about Samsung SIP Phones.

[CONFIGURATION > Miscellaneous > License]

### Multi-Extension Phone Configuration

Configure the information about Multi-Extension Phone.

[CONFIGURATION > User > Multi-Extension Phone]

- 1) Select the 'Phone Type' to 'Samsung-Desktop-Phone'.
- 2) 'Phone Name' and 'Profile Login ID' are not duplicated with the other user's configuration.
- 3) Depending on the way using phone can be set 'User Type', 'Phone Verification'. (Refer to Operation Manual)

## Multi-Phone User Configuration

Configure the information about Multi- Phone User.

[CONFIGURATION > User > Multi-Phone User]

- 1) 'Extension Number' and 'Application User ID' are not duplicated with the other user's configuration.
- 2) Select the phone name to 'Phone' menu among 'Multi-Extension Phone' configured above.
- 3) In Active-Active System case, 'A-A Primary Node' must be set.

## Phone Configuration

Select a 'Configure type' to 'Server' in Easy Install menu of phone.

Enter 'Profile Login ID' in Single Phone User menu into 'Login ID' in Easy Install menu of phone.

Enter 'Profile Login Passcode' in Single Phone User menu into 'Password' in Easy Install menu of phone.

Enter 'SCM IP Address' into 'Config Server' in Easy Install menu of phone.

## Registration Checking

You can check a registration status of phone in next menu.

[CONFIGURATION > Registration Status > Registration Status]

## 1.4 Multi-Extension Phone (PNP Mode)

### License Checking

Check a license count about Samsung SIP Phones.

[CONFIGURATION > Miscellaneous > License]

### Multi-Extension Phone Configuration

Configure the information about Multi-Extension Phone.

[CONFIGURATION > User > Multi-Extension Phone]

- 1) Select the 'Phone Type' to 'Samsung-Desktop-Phone'.
- 2) 'Phone Name' and 'Profile Login ID' are not duplicated with the other user's configuration.
- 3) Select the 'Phone Verification' to 'MACAddress' and enter the MAC address of phone into 'MAC Address'.
- 4) Depending on the way using phone can be set 'User Type', 'Phone Verification'. (Refer to Operation Manual)

## Multi-Phone User Configuration

Configure the information about Multi- Phone User.

[CONFIGURATION > User > Multi-Phone User]

- 1) 'Extension Number' and 'Application User ID' are not duplicated with the other user's configuration.
- 2) Select the phone name to 'Phone' menu among 'Multi-Extension Phone' configured above.
- 3) In Active-Active System case, 'A-A Primary Node' must be set.

## DHCP Server Configuration

Add the next item to DHCP Server.

(Use a SCM IP Address instead of 1.1.1.1)

DHCP Option 43:

[A] code: 43      type: string      data: sec,tftp://1.1.1.1

## Phone Configuration

Select a 'Configure type' to 'PnP' in Easy Install menu of phone.

## Registration Checking

You can check a registration status of phone in next menu.

[CONFIGURATION > Registration Status > Registration Status]

## 2 Interworking local site 3<sup>rd</sup> party SIP phone

3rd-party SIP Phone can be used only Sing Phone User type.

### License Checking

Check a license count about 3rd Party SIP Phones.

[CONFIGURATION > Miscellaneous > License]

Field	Value	Field	Value
License Key Type	SCM Express - Users	License Key	6RNMFHUB-Z4FC2Y80-GYXQBMSM-USECHYMQ
MAC Address	000C29A9CF9A	License Status	OK
Samsung SIP Phones	100	Samsung Soft Phones	100
Samsung Mobile Phones	100	Samsung PC Attendants	100
3rd Party SIP Phones	100	Analog Phones(Gateway)	100
AA Availability(Master/Slave)	No	High Availability(Active/Standby)	No
Meet-Me Conference Channels		UMS Channels	
Total CSTA Applications		Samsung Operators	
Embedded ACD Agent Links		Communicators(Desktop)	
Other CSTA Applications		SIP Application Channels	
FMS Phones		mVoIP Phones	

### Single Phone User Configuration

Configure the information about Single Phone User.

[CONFIGURATION > User > Single Phone User]

Field	Value	Field	Value
User Group	UG1	Service Group	UG1-SG1
Location	UG1-LOC1	Extension Number	6001
Application User ID	6001@ug1.scm.com	Extension Name	6001
Application Password	samsung	PIN Number	0000
Authentication User ID	6001	Phone Verification	None
Authentication Password	0000	MAC Address	
IP Address	165.213.80.252	Private IP Address	165.213.80.252
Profile Login ID	UG16001	Phone Type	3rd-Party-SIP-Phone
Profile Login Passcode	0000	Language	English
Mobile Phone Number		Use Mobile Phone Number	None
Protocol	UDP	Media	RTP
TLS Connection	Reuse	Ping Ring Type	None
A-A Primary Node	NODE 0	A-A Dual Registration	Enable
VMS Extension Number		Make Mailbox	Yes

- 1) Select the 'Phone Type' to '3rd-Party-SIP-Phone'.
- 2) 'Extension Number', 'Application User ID' and 'Profile Login ID' are not duplicated with the other user's configuration.
- 3) Enter the 'Authentication User ID' and 'Authentication Password'.
- 4) In Active-Active System case, 'A-A Primary Node' must be set.

### Phone Configuration

Enter 'SCM IP Address', 'Authentication User ID' and 'Authentication Password' in 3rd Party SIP Phone.

### Registration Checking

You can check a registration status of phone in next menu.

[CONFIGURATION > Registration Status > Registration Status]

## 3 Interworking home worker desktop phone

Interworking home worker desktop phone is same with interworking local desktop phone basically.

Refer to interworking local desktop phone about basic setting.

Additional setting is as follows.

### 3.1 Case of using SBC

#### SBC Configuration

Check a license count about Samsung SIP Phones.

Next port must be forwarded to SCM in SBC.

- To SIP signaling: UDP (5060), TCP (5060), TCP (5061)
- To RTP packet: RTP port range (40000~40799)

Please refer to '1.12 Interworking SBC' page to detail setting.

#### Firewall Configuration

Public IP must be needed to access to SCM from home.

Next port must be forwarded to SCM in Firewall for using Public IP.

- To upgrade package of phone: UDP (6000)
- HTTP: TCP (80)
- HTTPS: TCP (443)
- TFTP Server: UDP (69), TCP (69)

Please refer to '1.13 NAT/Firewall configuration' page to detail setting.

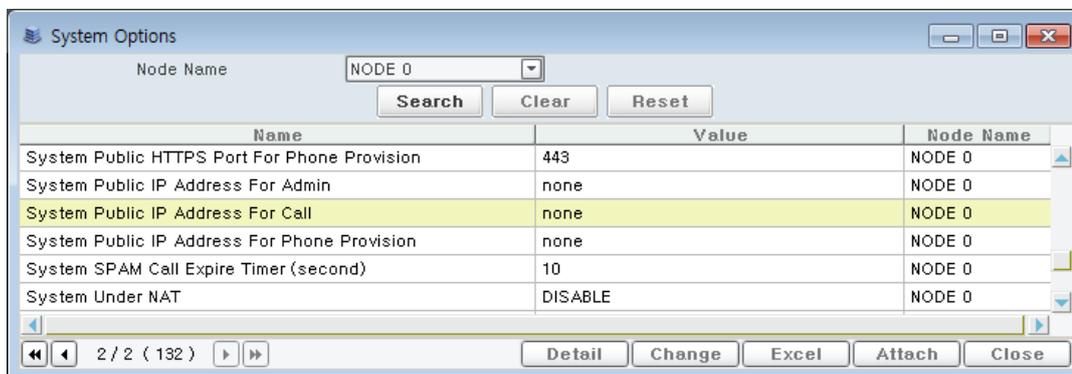
#### Phone Configuration

Enter 'Public IP Address' of firewall into 'Config Server' in Easy Install menu of phone.

## SCM Configuration

Next configuration is needed in next menu.

[CONFIGURATION > Miscellaneous > System Options]



- 1) Enter the Public IP of SBC into 'System Public IP Address For Call'.
- 2) Enter the Public IP of firewall into 'System Public IP For Phone Provision'.

## 3.2 Case without SBC

### Firewall Configuration

Public IP must be needed to access to SCM from home.

Next port must be forwarded to SCM in Firewall for using Public IP.

- To SIP signaling: UDP (5060), TCP (5060), TCP (5061)
- To RTP packet: RTP port range (40000~40799)
- To upgrade package of phone: UDP(6000)
- HTTP: TCP (80)
- HTTPS: TCP (443)
- TFTP Server: UDP (69), TCP (69)

Please refer to '1.13 NAT/Firewall configuration' page to detail setting.

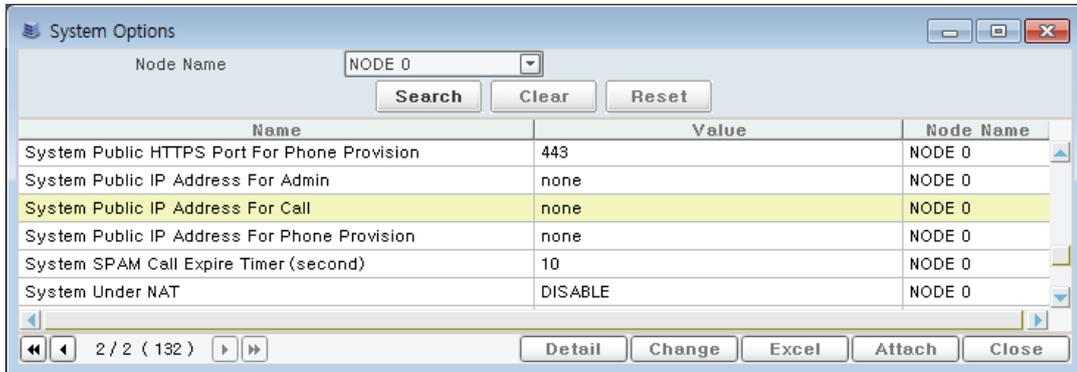
### Phone Configuration

Enter 'Public IP Address' of firewall into 'Config Server' in Easy Install menu of phone.

## SCM Configuration

Next configuration is needed in next menu.

**[CONFIGURATION > Miscellaneous > System Options]**



Name	Value	Node Name
System Public HTTPS Port For Phone Provision	443	NODE 0
System Public IP Address For Admin	none	NODE 0
System Public IP Address For Call	none	NODE 0
System Public IP Address For Phone Provision	none	NODE 0
System SPAM Call Expire Timer (second)	10	NODE 0
System Under NAT	DISABLE	NODE 0

- 1) Enter the Public IP of firewall into 'System Public IP Address For Call'.
- 2) Enter the Public IP of firewall into 'System Public IP For Phone Provision'.

## 4 Interworking remote site desktop phone

Interworking remote site desktop phone is divided as follow depend on network configuration.

### 4.1 Same Network Configuration

It is same with interworking local desktop phone basically.  
Please refer to '1 Interworking local site desktop phone' page.

### 4.2 Different Network Configuration

It is same with interworking home worker desktop phone basically.  
Please refer to '3 Interworking home worker desktop phone' page.

## 5 Interworking remote site 3<sup>rd</sup> party SIP phone

Interworking remote site 3<sup>rd</sup> party SIP phone is divided as follow depend on network configuration.

### 5.1 Same Network Configuration

It is same with interworking local site 3<sup>rd</sup> party SIP phone.  
Please refer to '2 Interworking local site 3<sup>rd</sup> party SIP phone' page.

### 5.2 Different Network Configuration

It is same with interworking local site 3<sup>rd</sup> party SIP phone basically.  
Please refer to '2 Interworking local site 3<sup>rd</sup> party SIP phone' page about basic setting.  
Additional setting is as follows.

#### 5.2.1 Case of using SBC

##### SBC Configuration

Check a license count about Samsung SIP Phones.  
Next port must be forwarded to SCM in SBC.

- To SIP signaling: SIP signaling port of 3<sup>rd</sup> Party Phone
- To RTP packet: RTP port range of 3<sup>rd</sup> Party Phone

Please refer to '12 Interworking SBC' page to detail setting.

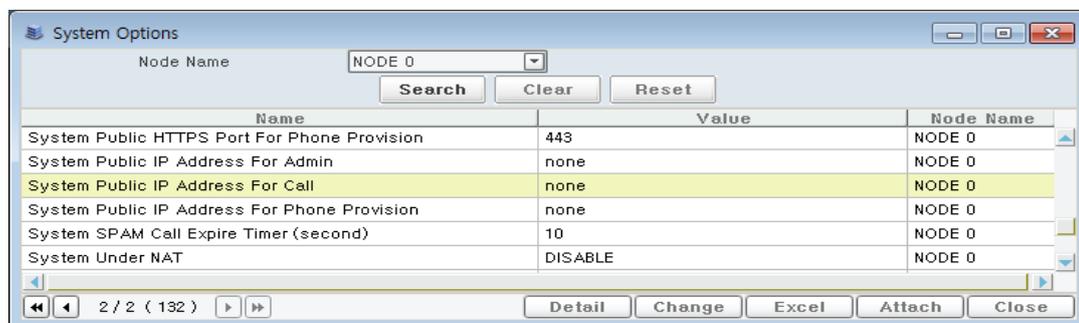
##### Phone Configuration

Enter 'Public IP Address' of SBC into 'Server IP' in 3<sup>rd</sup> party phone.

##### SCM Configuration

Next configuration is needed in next menu.

[CONFIGURATION > Miscellaneous > System Options]



Enter the Public IP of SBC into 'System Public IP Address For Call'.

## 5.2.2 Case without SBC

### Firewall Configuration

Public IP must be needed to access to SCM from home.

Next port must be forwarded to SCM in Firewall for using Public IP.

- To SIP signaling: SIP signaling port of 3<sup>rd</sup> Party Phone
- To RTP packet: RTP port range of 3<sup>rd</sup> Party Phone

Please refer to '13 NAT/Firewall configuration' page to detail setting.

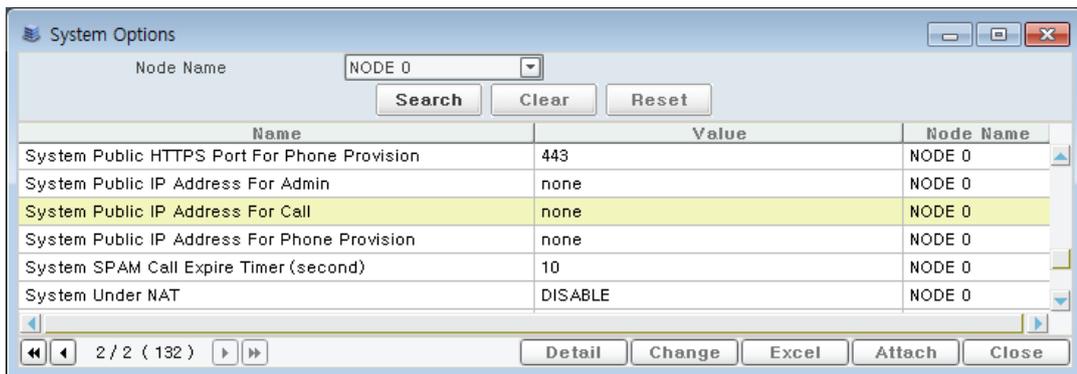
### Phone Configuration

Enter 'Public IP Address' of firewall into 'Server IP' in 3<sup>rd</sup> party phone.

### SCM Configuration

Next configuration is needed in next menu.

[CONFIGURATION > Miscellaneous > System Options]



Enter the Public IP of firewall into 'System Public IP Address For Call'.

## 6 Interworking local gateway

### 6.1 SCM

SCM configuration is as follows for interworking local gateway.

#### Route Configuration

Route must be created for interworking gateway.

**[CONFIGURATION > Trunk Routing > Route]**

Route Type	User Group	User Group	UG1
Route Name	Gateway	Location	UG1-LOC1
Access Number		Register Type	Receive REGISTER
Proxy Server	165,213,177,248	Port	5060
User Name	GATEWAY	Domain Name	
Authentication User Name	1q2w3e	Authentication Password	1q2w3e
DNS		Outbound CLI Prefix	
DTS Mode	Disable	A-A Primary Node	NODE 0
A-A Dual Registration	Enable		

- 1) Select the 'Register Type' to 'Receive REGISTER'.
- 2) Enter the Gateway IP into 'Proxy Server'.
- 3) Enter the 'User Name', 'Authentication User Name' and 'Authentication Password' to refer a gateway setting.
- 4) In Active-Active System case, 'A-A Primary Node' must be set.

## Gateway Link Setting

Create a Gateway Link in next menu.

[CONFIGURATION > Gateway > Gateway Link Setting]

The screenshot shows a 'Create' dialog for 'Gateway Link Setting'. The fields are as follows:

User Group	UG1	Name	GATEWAY
Gateway Type	OfficeServ 7400	IP Address(for Provision)	165.213.177.248
IP Address(for SIP register)	165.213.177.248	Public IP Address	
NAT	Disable	MAC Address(1)	
MAC Address(0)	11:22:33:44:55:66	Login Password(MAT)	
URL			
Login IP Address(MAT)			

Survivability Users - SIP: [ Selected ]

Survivability Users - FXS: [ Selected ]

Buttons: Create, Apply, Close

- 1) Name must be same with 'User Name' in Route.
- 2) Select the 'Gateway Type'.
- 3) Enter the Gateway IP into 'IP Address (for SIP register)'/ 'IP Address (for Provision)'.
- 4) Select a 'Disable' to 'NAT'.
- 5) Enter the MAC Address of Gateway into 'MAC Address (0)'

## Registration Checking

You can check a registration status of Gateway in next menu.

[CONFIGURATION > Registration Status > Registration Status]

## 6.2 OfficeServ7400

### IP setting

On the SIO screen connected to OfficeServ 7000 series, enter the 'ip\_help' command to view a list of available commands.

Set the IP address with the 'ip\_set' command, check the IP address with the 'ip\_config' command, and restart OfficeServ 7000 series with the 'sys\_reset' command.

```
-> ip_help

ip_help           Display this help message
ip_set "IP","NM","GW" Set IP Address, NetMask, Gateway
ip_config         Show Info. about Network Interfaces
sys_reset        System restart

-> ip_set "165.213.89.141", "255.255.255.0", "165.213.89.1"

>>> IP Address      = 165.213.89.141
>>> Subnet Mask     = 255.255.255.0
>>> Gateway         = 165.213.89.1

-> ip_config

          <<<<< Network Configuration >>>>>
+-----+-----+-----+-----+
| No | Items                | Value                |
+-----+-----+-----+-----+
| 0 | MAC Address          | 00-00-F0-E8-6F-AA   |
+-----+-----+-----+-----+
| 1 | IP Address           | 165.213.89.141      |
| 2 | Subnet Mask          | 255.255.255.0       |
| 3 | Default Gateway     | 165.213.89.1        |
+-----+-----+-----+-----+

-> sys_reset
```

### Country configuration

You should use DM from now.

DM 2.1.0 System Selection → System Country → select the country.

2.1.0.System Selection	
Item	Value
System Country	KOREA
IP Version	IPv4
IP Address	165.213.89.141
Gateway	165.213.89.1
Subnet Mask	255.255.255.0
WBS Select	Dual

### MGI configuration

DM 2.2.2-configure MGI IP.

2.2.2.MGI Card	
C1-S5	
Item	Value
Card Type	MGI 16/64
IP Address	10.254.168.40
Gateway	10.254.168.1
Subnet Mask	255.255.255.0
IP Type	Private Only
MAC Address	00:16:32:C5:A3:03
Local RTP Port (start)	30000
Public IP Address 1	0.0.0.0
Public RTP Port 1	30000
Public IP Address 2	0.0.0.0
Public RTP Port 2	30000
Public IP Address 3	0.0.0.0
Public RTP Port 3	30000

## Provisioning Link configuration

You should configure the Provisioning Link to receive the User Profile data from the SCM Express

DM 5.6.1-set the SCM IP at Master IP address

If SCM is Active-Active configuration, set the Slave SCM IP at Slave IP Address

5.6.1. System I/O Parameter		
Item		Value
MGI Alive Time (sec)		5
SCM Express Server	Master IP Address	165.213.66.93
	Slave IP Address	255.255.255.255

## Connecting to SCM Server

To have OfficeServ 7000 Series interoperate with SCM, the following items must be configured for SCM and OfficeServ 7000 Series

### 1) DM 5.2.13 SIP Carrier Options settings

Name	Description
SIP Carrier Name	Set to SCM
SIP server Enable	Enabling SIP server.
Outbound Proxy	Specify the IP address or the domain name of SCM.
Alternative Outbound Proxy	Specify the IP address of Slave SCM on Active-Active configuration.
Proxy Domain Name	Domain Name of SCM User Group.
User Name	Specify the number for the gateway to be registered as an endpoint.
Auth Username	Username of the gateway for authentication.
Auth Password	Password of the gateway for authentication.
Regist. Per User	Set to Per User
Trunk CLI Table	Specify a table containing the called phone number to be used for incoming analog trunk calls. Select one from Tables 1 through 4 in the 'DM 2.4.3 Send CLI Number'
Dual Registration	Set to Enable on Active-Active configuration

5.2.13.SIP Carrier Options	
SIP Carrier <input type="text" value="1"/>	
Item	Value
SIP Carrier Name	SCM
SIP Server Enable	Enable
SIP Service Available	Yes
Registra Address	
Registra Port	5060
Outbound Proxy	165.213.66.93
Alternative Outbound Proxy	0.0.0.0
Outbound Proxy Port	5060
Proxy Domain Name	ug1.scm.com
Local Domain Name	
SMS Domain Name	
DNS Server 1	0.0.0.0
DNS Server 2	0.0.0.0
User Name	OS7400
Auth Username	OS7400
Auth Password	*****
Regist Per User	Per User
Session Timer	None
Session Expire Time (sec)	1800
Trunk Reg Expire Time (sec)	1800
Representative Reg Expire Time (sec)	60
Alive Notify	None
Alive Notify Time (sec)	60
IMS Option	Disable
P Asserted ID Use	None
SIP Peering	Disable
Send CLI Table	1
Supplementary Type	PBX Managed 2
302 Response	Disable
SIP Destination Type	To Header
Codec Auto Nego	Enable
URI Type	SIP
SIP Signal Type	UDP
PRACK Support	Disable
Hold Mode	Send Only
Response to Tag	Keep
SIP Connection Reuse	Enable
SIP Mutual TLS Enable	Enable
SIP Validate Any TLS Certificate	Enable
SIP Trunking Codec PR1	G.729
SIP Trunking Codec PR2	G.711a
SIP Trunking Codec PR3	G.711u
SIP Trunking Codec PR4	Disable
SIP Trunking Use Alias	Disable
SIP Trunking Max Channel	224
Outgoing Originator Codec Use	Disable
Incoming Call Fixed Codec	Disable
Anonymous Host Name	Disable
Dual Registration	Disable
Trunk Ring Plan CLI Table	1
Trunk Group Number interworking SCM	805

## 2) DM 5.2.14 SIP User settings

Name	Description
User Name	Specify the number for the gateway to be registered as an endpoint.
Auth Username	Username of the gateway for authentication.
Auth Password	Password of the gateway for authentication.
Tel number	Enter the called group number for the trunk. The called group number is the number specified for 'Group Number' in the '4.1.2 Trunk Groups' menu of OfficeServ DM.

5.2.14.SIP Users						
Table No 1						
Entry No	User Name	Auth User Name	Auth Password	Tel Number	Registration Status	Registration Status 2nd
1	OS7400	OS7400	*****	805	Yes	No
2					No	No
3					No	No
4					No	No

## 6.3 iBG

### IP address & static route configuration

You need to connect to Ubigate iBG via Console with terminal program.

Following steps show the way to configure IP address.

```
iBG# configure terminal
iBG/configure# interface ethernet 0/1
iBG/configure/interface/ethernet (0/1)# ip address 10.10.10.100/24
iBG/configure/interface/ethernet (0/1)# exit
iBG/configure# ip route 0.0.0.0/0 10.10.10.1
```

### SCM connection

Following example shows the way to configure voip gateway.

```
iBG# configure terminal
iBG/configure# voip-gateway

// VoIP gateway must be shutdown before setting.
iBG/configure/voip-gateway# shutdown

// setting domain name.
iBG/configure/voip-gateway# host domain-name ugl.scm.com

// setting source interface for SIP and media.
iBG/configure/voip-gateway# bind control interface ethernet 0/1
iBG/configure/voip-gateway# bind media interface ethernet 0/1
iBG/configure/voip-gateway# call-server
```

```
// Setting a SCM server IP Address.
iBG/configure/voip-gateway/call-server# ip-address ipv4:10.10.10.10
iBG/configure/voip-gateway/call-server# ip-address ipv4:10.10.10.11
secondary

// Setting gw-uri and Ubigate iBG will register to SCM as an Endpoint.
iBG/configure/voip-gateway/call-server# gw-uri ibg-gw-001
iBG/configure/voip-gateway/call-server# exit

// Setting the username and password for G/W endpoint authentication.
iBG/configure/voip-gateway# sip-ua authentication username ibg-name
password pw1234

// Restarts VoIP gateway.
iBG/configure/voip-gateway# no shutdown
iBG/configure/voip-gateway# exit
iBG/configure# exit

// save current configuration.
iBG# save local
```

## 7 Interworking remote site gateway

### 7.1 SCM

SCM configuration is as follows for interworking remote site gateway.

#### 7.1.1 Same Network Configuration

It is same with interworking local gateway basically.  
Please refer to '6 Interworking local gateway' page.

#### 7.1.2 Different Network Configuration (Gateway in Public Network)

##### Route Configuration

Route must be created for interworking gateway.

[CONFIGURATION > Trunk Routing > Route]

Route Type	User Group	User Group	UG1
Route Name	Gateway	Location	UG1-LOC1
Access Number		Register Type	Receive REGISTER
Proxy Server	165,213,177,248	Port	5060
User Name	GATEWAY	Domain Name	
Authentication User Name	1q2w3e	Authentication Password	1q2w3e
DNS		Outbound CLI Prefix	
DTS Mode	Disable	A-A Primary Node	NODE 0
A-A Dual Registration	Enable		

- 1) Select the 'Register Type' to 'Receive REGISTER'.
- 2) Enter the Public IP of Gateway into 'Proxy Server'.
- 3) Enter the 'User Name', 'Authentication User Name' and 'Authentication Password' to refer a gateway setting.
- 4) In Active-Active System case, 'A-A Primary Node' must be set.
- 5) If there are NAT instead of SBC between SCM and Gateway, Select a 'Enable' to 'Nat Traversal' Option.

## Gateway Link Setting

Create a Gateway Link in next menu.

[CONFIGURATION > Gateway > Gateway Link Setting]

- 1) Name must be same with 'User Name' in Route.
- 2) Select the 'Gateway Type'.
- 3) Enter the Gateway IP into 'IP Address (for SIP register)'/ 'IP Address (for Provision)'.
- 4) Select a 'Disable' to 'NAT'.
- 5) TCP 8088 port must be forwarded to SCM in Firewall for using Public IP.
- 6) Enter the MAC Address of Gateway into 'MAC Address (0)'

## Registration Checking

You can check a registration status of Gateway in next menu.

[CONFIGURATION > Registration Status > Registration Status]

### 7.1.3 Different Network Configuration (Gateway under NAT)

#### Route Configuration

Route must be created for interworking gateway.

[CONFIGURATION > Trunk Routing > Route]

- 1) Select the 'Register Type' to 'Receive REGISTER'.
- 2) Enter the Public IP of Gateway into 'Proxy Server'.
- 3) Enter the 'User Name', 'Authentication User Name' and 'Authentication Password' to refer a gateway setting.
- 4) In Active-Active System case, 'A-A Primary Node' must be set.
- 5) If there are NAT instead of SBC between SCM and Gateway, Select a 'Enable' to 'Nat Traversal' Option.

#### Gateway Link Setting

Create a Gateway Link in next menu.

[CONFIGURATION > Gateway > Gateway Link Setting]

- 1) Name must be same with 'User Name' in Route.
- 2) Select the 'Gateway Type'.
- 3) Enter the Gateway real IP into 'IP Address (for SIP register)'/ 'IP Address (for Provision)'.
- 4) Select a 'Enable' to 'NAT'.
- 5) Enter the Public IP of Gateway into 'Public IP Address'.
- 6) TCP 8088 port must be forwarded to SCM in Firewall for using Public IP.
- 7) Enter the MAC Address of Gateway into 'MAC Address (0)'

### Registration Checking

You can check a registration status of Gateway in next menu.

[CONFIGURATION > Registration Status > Registration Status]

## 7.2 OfficeServ7400

It is same configuration with local gateway case except for one setting. If SBC exist between SCM and remote site gateway, SCM IP address must be configured with SBC's IP address.

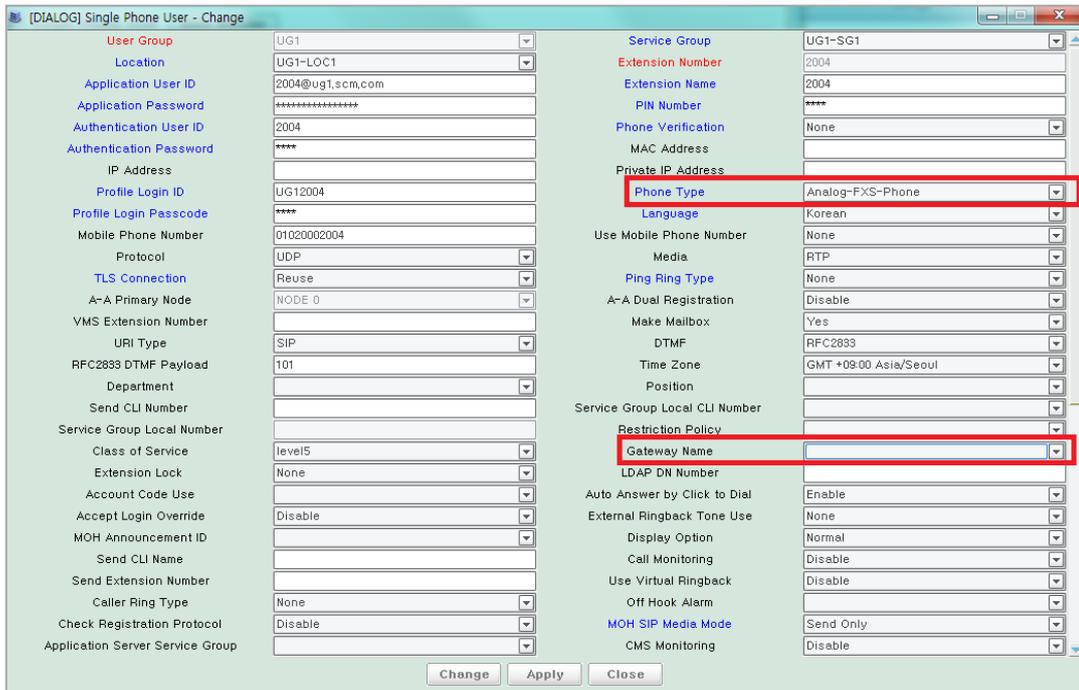
## 7.3 iBG

It is same configuration with local gateway case except for one setting. If SBC exist between SCM and remote site gateway, call-server IP address must be configured with SBC's IP address.

# 8 Interworking FXS

## 8.1 SCM

Create a user on the phone type set to Analog-FXS-Phone and, if necessary, to operate in survival mode by specifying the gateway will generate.



## 8.2 OfficeServ7400

### FXS Settings

- 1) DM 5.2.14 SIP User settings

Item	Description
User Name	Specify the number for the extension to be registered as a subscriber.
Auth Username	ID of the extension for authentication.
Auth Password	Password of the extension for authentication.
Tel number	Specify the extension number of '2.8.0 Numbering Plan' of OfficeServ DM

ex) FXS 3007 registration

5.2.14.SIP Users						
Table No <b>1</b>						
Entry No	User Name	Auth User Name	Auth Password	Tel Number	Registration Status	Registration Status 2nd
1	OS7400	OS7400	*****	805	Yes	No
2	3007	3007	****	3007	Yes	No
3					No	No
4					No	No

- 2) DM 2.4.3 Send CLI Number settings

Enter the FXS phone's extension number in the table selected for 'Send CLI Table' in the '5.2.13 SIP Carrier Options' menu of OfficeServ DM.

2.4.3.Send CLI Number					
Tel Number	Send CLI Number				Sen
	1	2	3	4	
2033					
2034					
2035	3007				
2036					
2037					

### 8.3 iBG

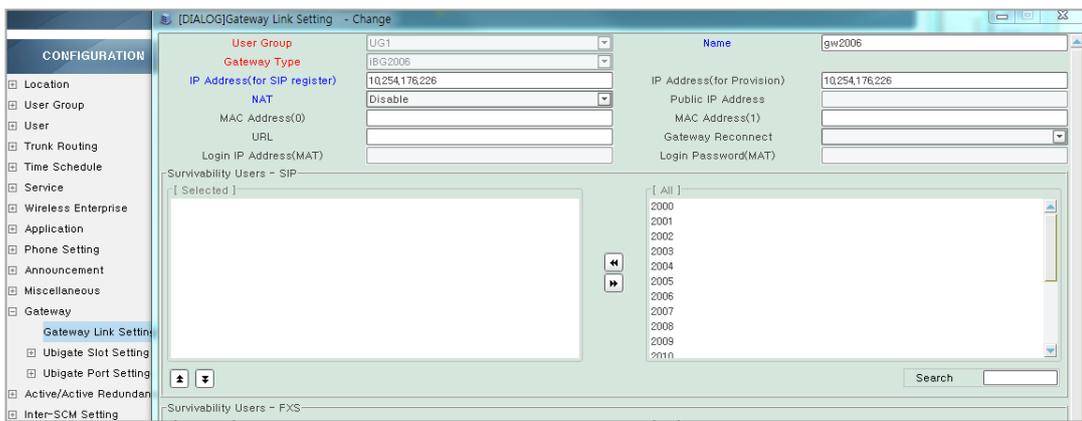
#### MAC address Setting

Following steps show the way to get system MAC address of iBG.

```
iBG# show mac system
Starting MAC = 00:16:32:xx:xx:xx
```

Setting the address in following menu.

#### [Gateway > Gateway Link Setting]

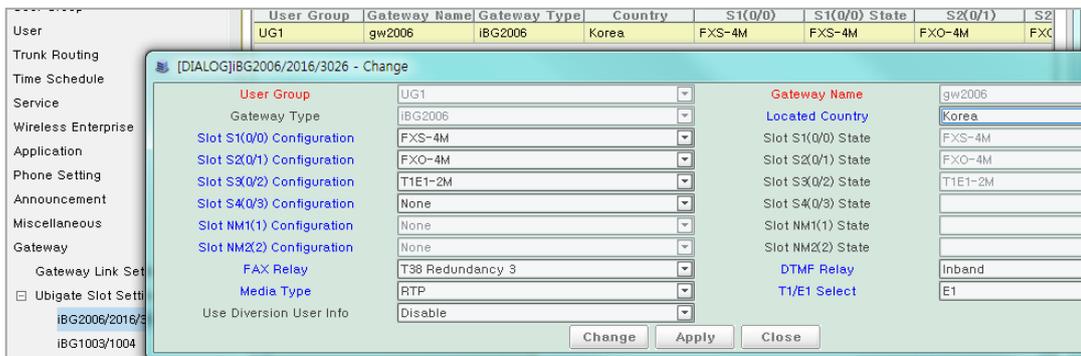


#### Slot Setting

Select a Gateway which you want to configure in the following window, and press [Change] button and select change.

Select a Slot Configuration item and choose a card type properly. Slot State indicates actual card equipment state. This state can be updated by receiving information from the gateway.

#### [Gateway > Ubigate Slot Setting > iBG2006/2016/3026] or [Gateway > Ubigate Slot Setting > iBG1003/1004]



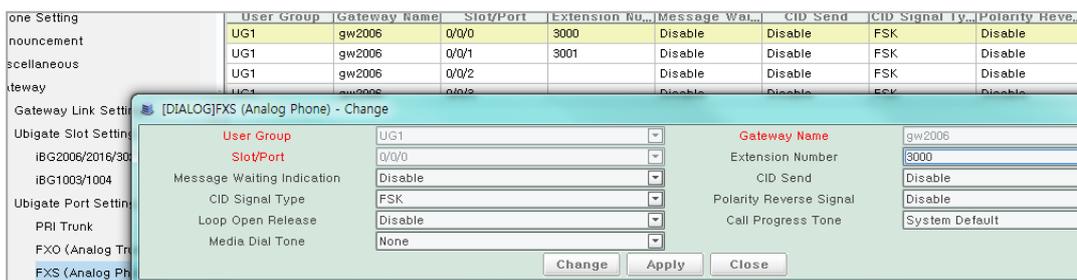
### Port Setting

Select an FXS port which you want to configure in the following window, and press **[Change]** button and select change.

To set extension number of an FXS port, click the 'Extension Number' menu and select a number from listed numbers. To configure this, FXS user must be configured in advance.

To remove this port configuration, select the blank instead of the number.

#### [Gateway > Ubigate Port Setting > FXS (Analog Phone)]



## 9 Interworking PRI

### 9.1 SCM

#### Creating Route

Route must be created for interworking gateway.

**[CONFIGURATION > Trunk Routing > Route]**

- 1) Select the 'Register Type' to 'Receive REGISTER'.
- 2) Enter the Public IP of Gateway into 'Proxy Server'.
- 3) Enter the 'User Name', 'Authentication User Name' and 'Authentication Password' to refer a gateway setting.
- 4) In Active-Active System case, 'A-A Primary Node' must be set.
- 5) Fill out Access Number to select The PRI trunk

#### Selecting Access Code Type and Setting DID number

**[CONFIGURATION > Trunk Routing > Route > Access Code]**

If you want to send number with access code, select the internal number type.

Normal Type make number to send without access number.

**[CONFIGURATION > Trunk Routing > Route > DID Routing]**

If you want receive call from PRI, you must configure DID number.

## 9.2 OfficeServ7400

### PRI settings

DM 5.2.14 SIP User setting

Item	Description
User Name	Specify the name for the PRI trunk group to be registered as an endpoint.
Auth Username	ID of the endpoint for authentication.
Auth Password	Password of the endpoint for authentication.
Tel number	Enter the called group number for the PRI trunk. The called group number is the number specified for 'Group Number' in the '4.1.2 Trunk Groups' menu of OfficeServ DM.

ex) PRI registration (OS7400\_pri)

5.2.14.SIP Users						
Table No 1						
Entry No	User Name	Auth User Name	Auth Password	Tel Number	Registration Status	Registration Status 2nd
1	OS7400	OS7400	*****	805	Yes	No
2	OS7400_pri	OS7400_pri	*****	806	Yes	No
3					No	No
4					No	No

## 9.3 iBG

### Slot setting

Refer to Interworking FXS for slot setting. T1E1 card is required for PRI.

### Port setting

Select an ISDN-PRI trunk which you want to configure in the following window, and press [Change] button and select change. Change configurations of the ISDN-PRI trunk, in this window you can change items.

To select name of an ISDN-PRI trunk, click 'Route Name' menu and select a route name from listed names. Route must be configured in advance. To remove this port configuration, select the blank instead of the name.

### [Gateway > Ubigate Port Setting > PRI Trunk]

User Group	Gateway Name	Slot/Port	Route Name	Sending Com.	PRI Trunk Type	Switch Type	Use Channels
UG1	gw2006	Q/2/0		Disable	TE	NI2	30

User Group	UG1	Gateway Name	gw2006
Slot/Port	Q/2/0	Route Name	
Sending Complete	Disable	PRI Trunk Type	TE
Switch Type	NI2	Use Channels	30

Change Apply Close

## 10 Interworking FXO

### 10.1 SCM

Route must be created for interworking gateway.

[CONFIGURATION > Trunk Routing > Route]

Route Type	User Group	User Group	UG1
Route Name	Gateway	Location	UG1-LOC1
Access Number		Register Type	Receive REGISTER
Proxy Server	165.213.177.248	Port	5060
User Name	GATEWAY	Domain Name	
Authentication User Name	1q2w3e	Authentication Password	1q2w3e
DNS		Outbound CLI Prefix	
DTS Mode	Disable	A-A Primary Node	NODE 0
A-A Dual Registration	Enable		

- 1) Select the 'Register Type' to 'Receive REGISTER'.
- 2) Enter the Public IP of Gateway into 'Proxy Server'.
- 3) Enter the 'User Name', 'Authentication User Name' and 'Authentication Password' to refer a gateway setting.
- 4) In Active-Active System case, 'A-A Primary Node' must be set.
- 5) Fill out Access Number to select The FXO trunk

## 10.2 OfficeServ7400

### FXO settings

- 1) DM 5.2.14 SIP User settings

Item	Description
User Name	Specify the name for the FXO trunk group to be registered as an endpoint.
Auth Username	ID of the endpoint for authentication.
Auth Password	Password of the endpoint for authentication.
Tel number	Enter the called group number for the FXO trunk. The called group number is the number specified for 'Group Number' in the '4.1.2 Trunk Groups' menu of OfficeServ DM.

ex) FXO registration (OS7400\_fxo)

5.2.14.SIP Users						
Table No 1						
Entry No	User Name	Auth User Name	Auth Password	Tel Number	Registration Status	Registration Status 2nd
1	OS7400	OS7400	*****	805	Yes	No
2	OS7400_fxo	OS7400_fxo	*****	8	Yes	No
3					No	No
4					No	No

- 2) DM 2.4.3 Send CLI Number setting

Enter the called phone number to use for incoming FXO trunk calls in the table selected for 'Trunk CLI Table' in the '5.2.13 SIP Carrier Options' menu of OfficeServ DM

2.4.3.Send CLI Number					
Tel Number	Send CLI Number				Send S
	1	2	3	4	
7001			3010		
7002					
7003					
7004					

## 10.3 iBG

### Slot setting

Refer to Interworking FXS for slot setting.

### Port setting

Select an FXO trunk which you want to configure in the following window, and press **[Change]** button and select change. Change configurations of the FXO trunk, in this window you can change items.

To select name of an FXO trunk, click 'Route Name' menu and select a route name from listed names. Route must be configured in advance. To remove this port configuration, select the blank instead of the name.

**[Gateway > Ubigate Port Setting > FXO (Analog Trunk)]**



# 11 Interworking SIP trunk

This section describes a process of creating information for trunks. The information shown below are the mandatory requirements for trunk calls, listed in the required order of creation.

- 1) Route information: This is the trunk port information for external connection to ITSP SIP servers, gateways, and other entities that interoperate with SCM.
- 2) LCR information: Specifying the preferred routes that are connected to the endpoint allows automatic selection of alternative routes and other factors.
- 3) LCR by location table information: You can specify different call routes according to the caller's location.
- 4) Access code information: This is the access code used for selecting trunk call routes.

## 11.1 Making Routes

A route means a conceptual path connected to an SIP server, a gateway, and other entities interoperating with SCM. The route includes information on the handling method for outgoing/incoming calls from each external connection endpoint as well as the number translation policy information.

You can create routes using the [CONFIGURATION > Trunk Routing > Route] menu. The blue text items are mandatory.

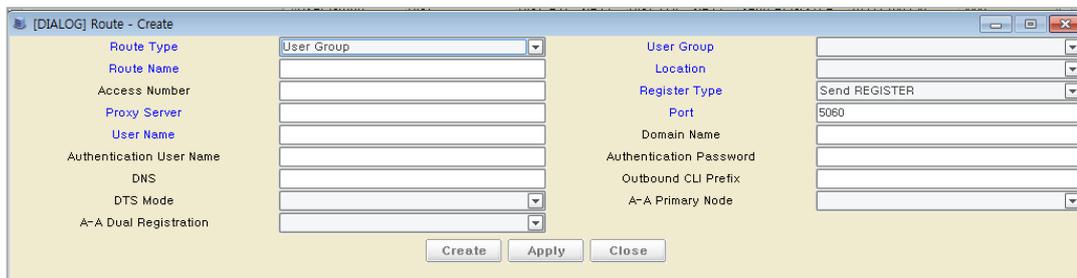


Figure 1. Making Routes

Item	Description
Route Type	Specify whether the route is used for one particular user group or shared by all user groups. - Common: Can be used by all user groups. If used for incoming calls only, additional settings are required, as number analysis is necessary to identify the user group being called. - User Group: Can be used only by one particular user group.
User Group	Select a user group to which the route belongs. If route type is set to Common, the user group also must be set to Common.
Location	Specify a location to which the endpoint belongs.

Item	Description
Register Type	Specify a registration method for the endpoint. - Receive REGISTER: REGISTER is received from the endpoint for registering the endpoint. - Send REGISTER: SCM sends REGISTER to the endpoint for registration. - None: No registration is performed between the endpoint and SCM.
Proxy Server	Specify the primary proxy server address for the endpoint.
Port	Specify a port number for the endpoint.
User Name	Specify the user name to use in user info of SIP URI.
DTS Mode	Specify the trunk option to use DTS Service
A-A Primary Node	Specify a node to use primary node. When trunk sends registration message, it sends to first selected node.
A-A Dual Registration	Specify the dual-registration. If register type is send trunk, this option must set 'disable'.
Access Code	Enter an access code to use when making calls to trunks instead of extension numbers. If you set this option in this step, Priority Routing, Location Based Routing and Access Code are created automatically.

When creating a route, you didn't input data that saved default. If you want changing data then you can change data.

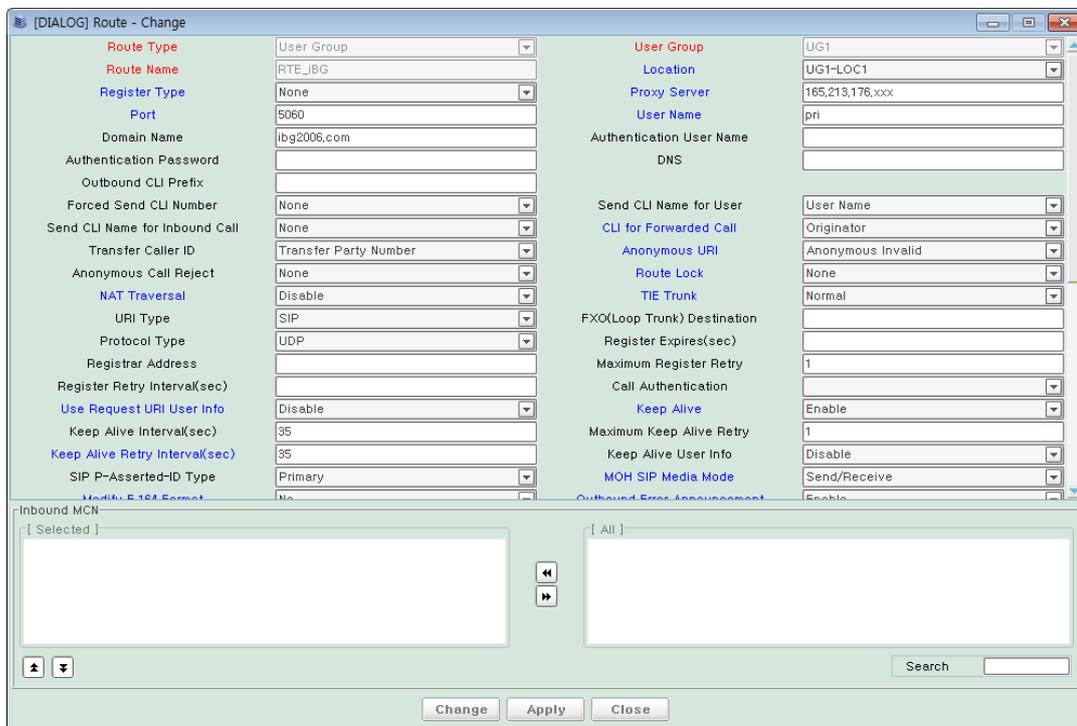


Figure 2. Modifying Routes

Item	Description
Domain Name	Enter a domain to use as the host of SIP URI.
Authentication User Name	Enter the user authentication name used for registration.
Authentication Password	Enter the authentication user password used for registration.
DNS	Enter the IP address of DNS server.
Outbound CLI Prefix	If there is no configuration of 'Send CLI Number' in a user and there is only extension number, when the user make a call through this route and there is prefix, add this prefix to the extension number and send it as calling number.
Forced Send CLI Number	When make a call to outbound, select the caller ID below: <ul style="list-style-type: none"> <li>- None: Follow the system priority. It is same as the order listed below.</li> <li>- Phone CLI Number: Send 'Phone CLI Number' as caller ID.</li> <li>- User CLI Number: Send 'User CLI Number' as caller ID.</li> <li>- Service Group CLI Number: Send 'Service Group CLI Number' as caller ID.</li> <li>- User Group CLI Number: Send 'User Group CLI Number' as caller ID</li> <li>- Outbound CLI Prefix + Extension Number: Send 'Outbound CLI prefix' with 'Extension Number' as caller ID.</li> <li>- Extension Number: Send 'Extension Number' as caller ID.</li> </ul>
Send CLI Name for User	When make a call to outbound, select the caller Name below: <ul style="list-style-type: none"> <li>- User Name: send 'User Name' as caller name.</li> <li>- Send CLI Name: Send 'Send CLI Name' as caller name.</li> </ul>
Send CLI Name for Inbound Call	If there is no caller name for inbound call, use below options. <ul style="list-style-type: none"> <li>- None: not used.</li> <li>- Receive CLI Number: use caller number as caller name.</li> </ul>
URI TYPE	Select SIPS if the protocol is TLS. Select SIP in other cases.
Protocol Type	Select UDP, TCP, or TLS as the protocol to use.
Register Expires (sec)	This is the expiration period for registration. SCM must retry registration within this period.
Register Retry Interval (sec)	Enter the interval for resending the REGISTER message.
Keep Alive	It is used to verify the connection using SIP OPTION message.
Keep Alive Interval (sec)	Specify an interval (seconds) for exchanging Keep Alive Messages
SIP P-Asserted-ID Type	Select a type of representative number. <ul style="list-style-type: none"> <li>- Primary: The P-Asserted-Identity header contains the Primary number and the From header contains residential number.</li> <li>- Secondary: The P-Asserted-Identity header contains the residential number and the From header contains Primary number.</li> </ul>
Modify E.164 Format	Specify whether to use E.164 format for calling number or called number for outgoing call through this route.
DNS SRV Query	Select whether to use of DNS SRV.
TLS Connection	Specify the TLS connection type:

Item	Description
	<ul style="list-style-type: none"> <li>- Normal: use TLS Full handshaking method.</li> <li>- Reuse: reuses the existing TLS connection established by initial Message.</li> <li>- Resume: use simplified handshaking method using TLS Session ID.</li> </ul>

## 11.2 Making LCR

A Least Cost Route (LCR) is a method of selecting a route when processing outgoing trunk calls. There are three different types of LCR, as shown below.

### Priority Routing

A priority routing allows automatic selection of alternative routes when the default outgoing path becomes unavailable. Priority is assigned to the direct route and alternative routes. When calls going out through the high-priority route fail, they can be retried through the low-priority routes.

You can create route sequences in the [**CONFIGURATION > Trunk Routing > Priority Routing**] menu. This menu is used for creating route sequences.

Item	Description
User Group	Specify a user group to which the route sequence belongs.
Name	Specify a name for the route sequence. Pay special attention to choosing the name, as it is used as an identifier when selecting the route sequence in other menus and cannot be changed.
Route Priority	Assign priority to the route. - Direct Route: Specify the top priority route. - Alternative Route1 to Alternative Route8: Select the routes according to their priority levels.
Route Name	Select a route for the route priority level.
Outbound DOD Delete Length	Specifies the length of digits to delete from the first position of the called number for outbound call.
Outbound DOD Insert Digits	Specifies the digits to insert from the first position of the called number for outbound call.
Outbound CLI Delete Length	Specifies the length of digits to delete from the first position of the calling number for outbound call.
Outbound CLI Insert Digits	Specifies the digits to insert from the first position of the calling number for outbound call.
Outbound MCN	Specify MCN rules for outbound call.

### Time-based Routing

A time-based routing sequence contains time conditions so that each service group can use different route sequences according to the conditions.

## Load-balanced Routing

A load-balanced routing allows use of the selected routes in a specified ratio.

Calls are distributed between the routes identified as available for calls, and therefore there is no need for configuring alternative routes.

Among the three types of LCRs, the route sequence type is used by default. For more information on LCRs, including the setup method and route selection, see the 'Least Cost Route (LCR) Policy' section of System Features.

## 11.3 Making Location Based Routing

A location based routing allows each location to use its own LCR. Since each location is set with its own LCR, if you have created multiple locations, you must set an LCR for each of the locations. If no LCR is set for a particular location, the location is not allowed to make trunk calls.

You can create route partitions in the **[CONFIGURATION > Routing > Location Based Routing]** menu. This menu is used for creating Location Based Routing.

Item	Description
User Group	Select a user group to which the Location Based Routing belongs.
Location Based Routing Name	Enter a name for the Location Based Routing. Pay special attention to choosing the name, as it is used as an identifier when selecting the route partition in other menus and cannot be changed.
Location select	Select whether to use the 'Location'. If Location selects use 'disable', all location can use this Location Based Routing.
Location	Select a location to set the routes.
Routing Type	Select a type of LCR for the location. - Time-Based Routing: Select to use a Time-Based Routing. - Priority Routing: Select to use a Priority Routing. - Load Balance Routing: Select to use a Load Balance Routing.

## 11.4 Configuring Access Codes

Access codes are number to be used for dialing a directly outgoing call. They are also used for analyzing the destination numbers to determine which location-based routes to be used for outgoing trunk calls.

You can create access codes using the [CONFIGURATION > Routing > Access Code] menu. The following items are mandatory.

Item	Description
User Group	Select a user group to which the access code belongs.
Access Code	Enter an access code to use when making calls to trunks instead of extension numbers. Pay special attention to choosing the code, as it is used as an identifier when selecting the access code in other menus and cannot be changed.
Number Type	Select a type of access code. The access code can be the beginning portion of the external destination number, or an internal code is used within the boundary of the SIP servers or gateways. - Normal: When the calling number for an outgoing trunk call is analyzed, the digit corresponding to the access code is deleted from the number dialed by the user, and then the call is made to the trunk. - Internal: When the calling number for an outgoing trunk call is analyzed, the digit corresponding to the access code is not deleted from the number dialed by the user and the call is made to the trunk as is. - Emergency: When the calling number for an outgoing trunk call is analyzed, only the digit corresponding to the access code from the number dialed by the user is used as the destination number, and then the call is made to the trunk. - Pattern: When analyzing the calling numbers for outgoing trunk calls, a wild card (expressed as X) is used to denote the length. The call is made to the trunk without deleting the digit corresponding to the access code from the number dialed by the user. - DTS: When the calling number for an outgoing trunk call is analyzed, the digit corresponding to the access code is not deleted from the number dialed by the users and the call is made to the trunk as is. DTS Access code can use DTS Trunk, other Trunk is not allowed.
Location Based Routing Name	Select a Location Based Routing to use with this access code.

## 11.5 Configuring DID Routing

Destination of incoming trunk calls depend on the DID number. When entering a DID number, you can use wild cards (entered by \*) to enter multiple numbers at a time. If the called number is set to 'B', a translated DID number is used as the called number.

Also the called number is set to 'E', a translated DID number is used as the called number. Digit differences allow tandem call or not.

You can specify different called numbers for different times of the day. Time periods are defined by ring plans.

For more information on DID number translation and assigning called numbers by ring plans, see the 'DID Routing' section of System Features. Also, for more information on ring plans, see the 'Ring Plan' section of System Features.

You can assign routes by DID number using the [CONFIGURATION > Routing > DID Routing] menu. The following items are mandatory.

Item	Description
User Group	Select a user group to which the calls are directed.
DID Number	Enter a DID number for incoming trunk calls.
Default Destination	Specify a called number to which the incoming calls with the selected DID number are directed. The default called number is used if the current ring plan is not RP1 through RP10.

## 12 Interworking SBC

This section describes a way to set iBG-SBC as sequence of installation. This section includes commands used generally for quick installation. Some setup method for setting network (for example, IP address, default gateway or TLS key) are not described here. You can see them in other sections for 'iBG' in this document. Note that this document contains commands for quick installation only. You can get more information for setup in 'configuration guide for iBG-SBC'.

### 12.1 Media-pool

You can setup a media-pool in iBG-SBC to pass media packets between public network and local network.

#### <Mandatory>

- 1) Set IP address to be used when media packets are passed.
- 2) Set port range to be used when media packets are passed.

#### <Setup method>

- 1) Create or delete media-pool: You can create or delete media-pool with command 'media-pool'. A name 'mpublic' below is just example. A name of media-pool is designated by operator.

```
SBC# configure terminal
SBC/configure# session-router
SBC/configure/session-router# media-pool mpublic (creation)
SBC/configure/session-router# no media-pool mpublic (deletion)
```

- 2) Setup IP address: You can set IP address to be used when media packets are passed with command 'ip-address'. A IP address '211.123.123.123' below is just example.

```
SBC/configure/session-router/media-pool mpublic# ip-address
211.123.123.123
```

- 3) Setup port range: You can set port range to be used when media packets are passed with command 'media-port'. First parameter (40000 in below) means starting value of port. Second parameter (512 in below) means range to be used. Range '40000-40512' in below is just example.

```
SBC/configure/session-router/media-pool mpublic# media-port 40000 512
```

## 12.2 Region

In iBG-SBC, logical network zone for classifying local and public side is called with 'region'. You can set physical network interface to region with commands below.

### <Mandatory>

- 1) Designate media-pool to be used in region.
- 2) Designate ethernet interface to be used in region.

### <Main option>

- 1) NAT option in region.  
(You should set it if there are SIP entities maintained behind NAT)
- 2) Set whether or not iBG-SBC manages SIP registration.  
(You should set it if there are SIP entities registering via iBG-SBC)
- 3) Change signaling port number to be used in region.  
(If you don't set it, iBG-SBC will use default port number, UDP-5060, TCP-5060 and TLS-5061)

### <Setup method>

- 1) Create or delete region: You can create or delete region with command 'region'.  
A name 'public' below is just example. A name of region is designated by operator.

```
SBC# configure terminal
SBC/configure# session-router
SBC/configure/session-router# region public (creation)
SBC/configure/session-router# no region public (deletion)
```

- 2) Designate media-pool: You can designate media-pool to be used in region with command 'media-pool'. You should designate local media-pool to local region and public media-pool to public region.

```
SBC/configure/session-router/region public# media-pool mpublic
```

- 3) Designate network interface: You can designate network interface to be used in region with command 'bind-interface'. A interface 'ethernet 0/1' below is just example.  
You should designate interface you select.

```
SBC/configure/session-router/region public# bind-interface ethernet
0/1
```

- 4) Set NAT option: You can set NAT option in region with command 'nat-traversal'.  
If you set NAT option with 'always', iBG-SBC detects SIP entities managed behind NAT with SIP message and handle it properly.

If you set NAT option with 'none', iBG-SBC doesn't care information for NAT.  
If there are SIP entities managed behind NAT, you should set this option with 'always'.

```
SBC/configure/session-router/region public# sip-interface
SBC/configure/session-router/region public/sip-interface# nat-
traversal always
```

- 5) Set SIP registrar option: You can set SIP registrar option in region with command 'route-to-registrar'. If you set this command, iBG-SBC records username and IP address with SIP 'REGISTER' message and preferentially uses this list to route SIP call.

```
SBC/configure/session-router/region public# sip-interface
SBC/configure/session-router/region public/sip-interface# registrar
SBC/configure/session-router/region public/sip-interface/registrar#
route-to-registrar
```

- 6) Change signaling port: You can change SIP signaling port in region with command 'local-port'. If you don't set it, iBG-SBC will use SIP default port (UDP=5060, TCP=5060, TLS=5061). If there is no customer's request about it, you don't need to change these ports.

```
SBC/configure/session-router/region public# sip-interface
SBC/configure/session-router/region public/sip-interface#
local-port udp 25060 tcp 25060 tls 25061
```

## 12.3 Policy

iBG-SBC has policy list to be used for routing SIP call. A policy is consist of condition and target. If SIP call corresponds with condition of certain policy, the call will be routed to target of that policy. Each policy has priority. iBG-SBC preferentially searches policy having high priority. If you set route-to-route to region, iBG-SBC will search SIP entities registered with SIP 'REGISTER' before searching policy list.

### <Mandatory>

- 1) Set condition about source region.
- 2) Set target region.
- 3) Set target address (It can be IP or DNS address).

### <Main option>

- 1) Set condition about source IP address.  
(Set this option, if you want to accept special IP addresses)
- 2) Set priority.  
(You can set searching sequence with this)

**<Setup method>**

- 1) Create or delete policy: You can create or delete policy with command 'policy'.  
A name 'loc2pub' below is just example. A name of policy is designated by operator.

```
SBC# configure terminal
SBC/configure# session-router
SBC/configure/session-router# policy loc2pub (creation)
SBC/configure/session-router# no policy loc2pub (deletion)
```

- 2) Set condition about source region: You can set condition of policy about source region with command 'source-region'. If you don't set it, policy will be out from searching list.

```
SBC/configure/session-router/policy loc2pub# source-region local
```

- 3) Set target region: You can set target region of policy with command 'destination-region'. If a SIP call corresponds with condition, the call will be route to this region.

```
SBC/configure/session-router/policy loc2pub# routing-policy
SBC/configure/session-router/policy loc2pub/routing-policy#
destination-region public
```

- 4) Set target address: You can set target address of policy with command 'next-hop'. It can be IP or DNS address. A IP and DNS addresses below is just example. Additionally, you can set target port with this command too.

```
SBC/configure/session-router/policy loc2pub# routing-policy
SBC/configure/session-router/policy loc2pub/routing-policy#
next-hop ipv4:128.123.123.123
next-hop ipv4:128.123.123.123:7080
next-hop dns:sec.sip-server.com
next-hop dns:sec.sip-server.com:6070
```

- 5) Set condition about source IP address: You can set condition about source IP addresses of policy with command 'source-address'. If you don't set it, iBG-SBC doesn't care source IP address. You can set it with subnet-type and range-type. See examples below.

```
SBC/configure/session-router/policy loc2pub#
source-address 211.34.34.0/24
source-address 112.45.45.45/32
source-address 120.30.30.1-120.30.30.100
```

- 6) Set priority: You can set searching priority of policy with command 'priority'. It can be set from 0 to 99. default value is 50. A smaller one has higher priority.

```
SBC/configure/session-router/policy loc2pub# priority 30
```

## 12.4 Security

iBG-SBC supports security function to protect SIP entities don't have permission.

iBG-SBC can check permission with IP address and stations. iBG-SBC has 'station' list. In iBG-SBC, stations mean SIP entities that try to register via iBG-SBC and receive '200 OK' response for it. You can set permission for security to call message (INVITE) and normal message (non-INVITE) separately.

### <Mandatory>

- 1) Create peer-group.
- 2) Set permission.

### <Setup method>

- 1) Create or delete peer-group: You can create or delete peer-group with command 'peer-group'. A name 'pg\_good' below is just example. A name is designated by operator. Peer-group has IP (or IP range) list with command 'add'. You can use 'no add' command to delete IP (range) from peer-group. You can set IP, IP range and port to peer-group. See below examples.

<Create or delete peer-group >

```
SBC# configure terminal
SBC/configure# session-router
SBC/configure/session-router# peer-group pg_good (creation)
SBC/configure/session-router# no peer-group pg_good (deletion)
```

<Add or delete IP addresses>

```
SBC/configure/session-router/peer-group pg_good#
add 2.3.4.5
add 2.3.4.5-2.3.4.10
add 2.3.4.5:1000-2.3.4.5:3000 (adding IP)
SBC/configure/session-router#
no add 2.3.4.5 (deleting IP)
```

- 2) Set permission: You can set permission with commands call-permission and message-permission in policy. A call-permission is used to set permission about call (INVITE message) and a message-permission is used to set permission about normal message (non-INVITE message). Subjects of permission can be peer-group or stations.

## &lt;Set call-permission&gt;

```
SBC/configure/session-router# policy pub2loc
SBC/configure/session-router/policy pub2loc# call-permission
SBC/configure/session-router/policy pub2loc/call-permission#
permission allow station
permission allow peer-group pg_good
permission reject peer-group pg_bad
```

## &lt;Set message-permission&gt;

```
SBC/configure/session-router# policy pub2loc
SBC/configure/session-router/policy pub2loc# message-permission
SBC/configure/session-router/policy pub2loc/message-permission#
permission allow peer-group pg_good
permission reject peer-group pg_bad
```

## 12.5 Activating SBC

You can activate and deactivate SBC function with command 'shutdown'. All iBG-SBC's configuration items, except 'features', can be set when SBC function is deactivated only.

## &lt;Setup method&gt;

- 1) Activate SBC: You can activate SBC function in iBG with command 'no shutdown'.

```
SBC# configure terminal
SBC/configure# session-router
SBC/configure/session-router# no shutdown
```

- 2) Deactivate SBC: You can deactivate SBC function in iBG with command 'shutdown'.

```
SBC# configure terminal
SBC/configure# session-router
SBC/configure/session-router# shutdown
```

## 13 NAT/Firewall configuration

When recognizing that there is a network-aware application program behind the NAT device, the NAT traversal feature obtains the external IP address and performs port mapping in order to transfer the data from the external port of the NAT device to the internal port used by the application program. All this is done automatically. There is no need for the user to manually perform port mapping or use any other mechanism.

To facilitate use of this technology in Internet telephony, SCM provides the media proxy feature for securing communication paths between the IP phones on the NAT network and the systems and phones on the public IP network.

As such, when it is difficult for SCM to obtain a public IP address, it can use the port mapping configuration of the existing NAT system to obtain a private IP address for SCM itself and provide a reliable Internet telephony service with IP phones on the public IP network or IP phones on another NAT network.

To implement this technology, the port information below must be set to open for the NAT system, and port mapping must be configured for the NAT system.

Following is a list of ports must be open when the SCM is located under NAT.

Service	TCP Port	UDP Port	Description
General	20, 21	-	FTP Server
	22	-	Secure Shell
	23	-	Telnet
	80, 443	-	HTTP Web Server
	389		LDAP Server
	123	123	NTP
Provisioning	69	-	TFTP Server
	8088	-	Gateway Provisioning
	-	6000	Phone upgrade from Proprietary to SIP
NMS	-	161	SNMP Agent
Personal Management	8080, 9500	-	Personal Assistant for Call Service
	4002, 4003, 4004	-	Single Sign-On, PWP for UMS/Conference
System Management	20001, 20002, 20003, 20005, 20006	-	SCM Administrator
	5432	-	PostGRE DBMS connection
Call	5060, 5061	5060	SIP signaling
UMS	5080, 8624	5080	Call signaling for UMS

Service	TCP Port	UDP Port	Description
	-	14002~14130	RTP path for UMS
	25, 143, 993	-	Signaling for E-mail Server
	3681, 3683, 2001, 22001	-	Signaling for Outlook client
	2200	-	UMS File Server
Conference	3333	5090, 5098	Call signaling for Conference
	-	44000~49998	RTP path for Conference
MOH	-	35000~35999	RTP path for MOH/Announcement
MPS	-	40000~40799	RTP path for MPS (Media Proxy Service)
Others	6000~6127	-	CSTA link for each user group
	9050, 9052	-	PMS link
	9090, 9092,9094	-	Proprietary Application server link
	9000, 9002,5110	-	Voice Monitoring server link
	9010,9011	-	MVS client link
	18124,18126	-	mySingle link
	10306, 2300	-	CDR (Call Data Record)
	-	1812,1813	Radius server
	1122	-	Active-Active node TCP port

# 14 Interworking WE VoIP

## Creating User

Create a user on the phone type set to Samsung-Mobile-Phone. Fill in the mobile number and select the using type of mobile phone number.

**[CONFIGURATION > User > Single Phone User]**

User Group	UG1	Service Group	UG1-SG1
Location	UG1-LOC1	Extension Number	2001
Application User ID	2001@ug1.scm.com	Extension Name	2001
Application Password	*****	PIN Number	****
Authentication User ID	2001	Phone Verification	None
Authentication Password	****	MAC Address	
IP Address	10.251.191.164	Private IP Address	10.251.191.164
Profile Login ID	UG12001	Phone Type	Samsung-Mobile-Phone
Profile Login Passcode	****	Language	Korean
Mobile Phone Number		Use Mobile Phone Number	None
Protocol	UDP	Media	RTP
TLS Connection	Reuse	Ping Ring Type	None
A-A Primary Node	NODE 0	A-A Dual Registration	Enable
VMS Extension Number		Make Mailbox	Yes
URI Type	SIP	DTMF	RFC2833
RFC2833 DTMF Payload	101	Time Zone	GMT +09:00 Asia/Seoul
Department		Position	
Send CLI Number		Service Group Local CLI Number	
Service Group Local Number		Restriction Policy	
Class of Service		Gateway Name	
Extension Lock	None	LDAP DN Number	
Account Code Use		Auto Answer by Click to Dial	Enable
Accept Login Override	Disable	External Ringback Tone Use	None
MOH Announcement ID		Display Option	Normal
Send CLI Name		Call Monitoring	Disable
Send Extension Number		Use Virtual Ringback	Disable
Caller Ring Type	None	Off Hook Alarm	
Check Registration Protocol	Disable	MOH SIP Media Mode	Send Only
Application Server Service Group		CMS Monitoring	Disable

### Configuration Mobile Service Option

Set the SSID of the WiFi SSID and the device WiFi must be the same.

[CONFIGURATION > Wireless Enterprise > Mobile Service Options]

**[DIALOG]Mobile Service Options - Change**

**User Group** UG1

Remote Dial Public IP Address: \_\_\_\_\_

Mobile DISA Number: \_\_\_\_\_

Mobile VMS DISA Number: \_\_\_\_\_

WE Work Server IP Address: \_\_\_\_\_

WE Work Server Public IP Address: \_\_\_\_\_

WE VoIP CID Server IP Address: \_\_\_\_\_

WE VoIP CID Server Public IP: \_\_\_\_\_

WE Work Server Protocol: HTTP

WE VoIP CID Server Public Protocol: HTTP

Wait Call, Later Call: False

Auto Answer CLI Number: \_\_\_\_\_

Use 3G Call Only: No

Logo File Path: \_\_\_\_\_

**SSID** \_\_\_\_\_

Remote Dial Public Port: \_\_\_\_\_

Mobile DISA Code: \_\_\_\_\_

WE Work Server Port: 80

WE Work Server Public Port: 80

WE VoIP CID Server Port: 80

WE VoIP CID Server Public Port: 80

WE VoIP CID Server Protocol: HTTP

WiFi Band: Auto

Auto Answer Profile Number: \_\_\_\_\_

3G Call Prefix: \_\_\_\_\_

**24G Channel List**

CH 1     CH 2     CH 3     CH 4

CH 5     CH 6     CH 7     CH 8

CH 9     CH 10     CH 11     CH 12

CH 13

Selected All

**5G Channel List**

CH 36     CH 40     CH 44     CH 48

CH 149     CH 153     CH 157     CH 161

CH 185

Selected All

Change Apply Close

### Configuration Phone Upgrade

Specify Server Address to upgrade Phone.

[CONFIGURATION > Wireless Enterprise > Upgrade Mobile Software]

**[DIALOG]Upgrade Mobile Software - Change**

**User Group** UG1

Download Server 1 - IP Address: \_\_\_\_\_

Download Server 2 - IP Address: \_\_\_\_\_

Download Server 3 - IP Address: \_\_\_\_\_

Download Server 4 - IP Address: \_\_\_\_\_

Download Server 5 - IP Address: \_\_\_\_\_

Download Server 1 - File Folder: \_\_\_\_\_

Download Server 2 - File Folder: \_\_\_\_\_

Download Server 3 - File Folder: \_\_\_\_\_

Download Server 4 - File Folder: \_\_\_\_\_

Download Server 5 - File Folder: \_\_\_\_\_

Change Apply Close

### Configuration Mobile Phone Profile

[CONFIGURATION > Wireless Enterprise > Mobile Phone Profile]

**[DIALOG]Mobile Phone Profile - Change**

**User Group** \_\_\_\_\_

Mobile Phone Number: \_\_\_\_\_

Select Download Server: \_\_\_\_\_

**Roaming Trigger** \_\_\_\_\_

Roaming Scan Period: \_\_\_\_\_

Noise Suppression TX: \_\_\_\_\_

Echo Suppression: \_\_\_\_\_

Enable Swing Free TX: \_\_\_\_\_

Media Start Port: \_\_\_\_\_

Multiframe Enable: \_\_\_\_\_

TOS Media Value(DSCP): \_\_\_\_\_

JBC Threshold: \_\_\_\_\_

**Extension Number** \_\_\_\_\_

User Agent Info: \_\_\_\_\_

Version: \_\_\_\_\_

Roaming Delta: \_\_\_\_\_

Noise Suppression RX: AECM

Enable Swing Free RX: \_\_\_\_\_

Enable CNG: \_\_\_\_\_

Media End Port: \_\_\_\_\_

Multicast Enable: \_\_\_\_\_

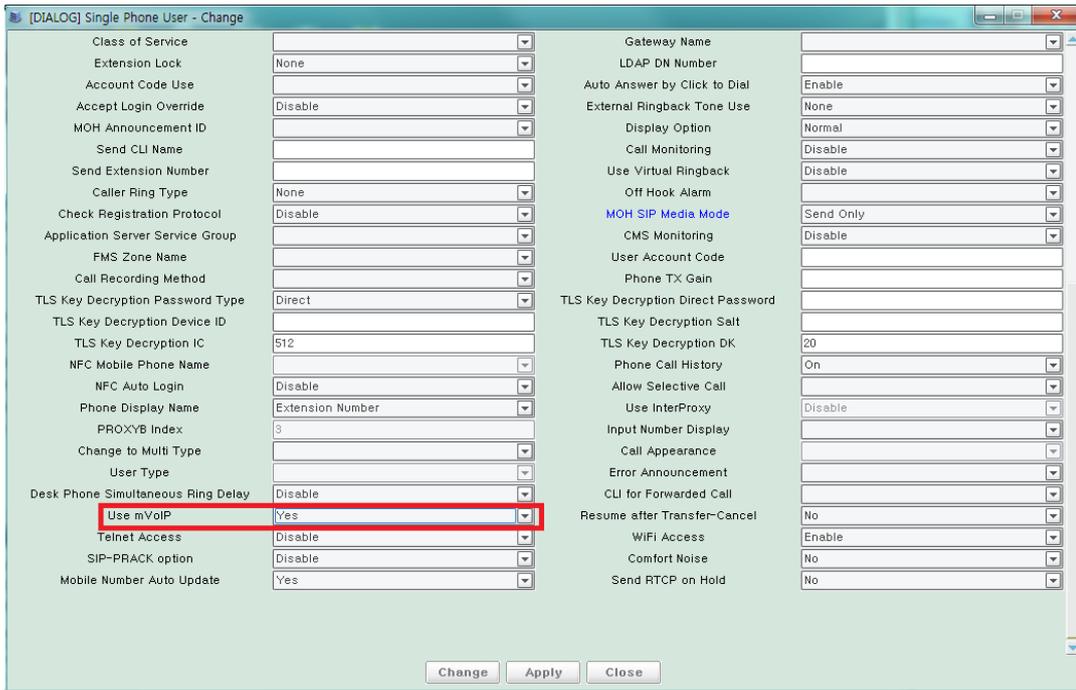
TOS Control Value(DSCP): \_\_\_\_\_

Change Apply Close

# 15 mVoIP

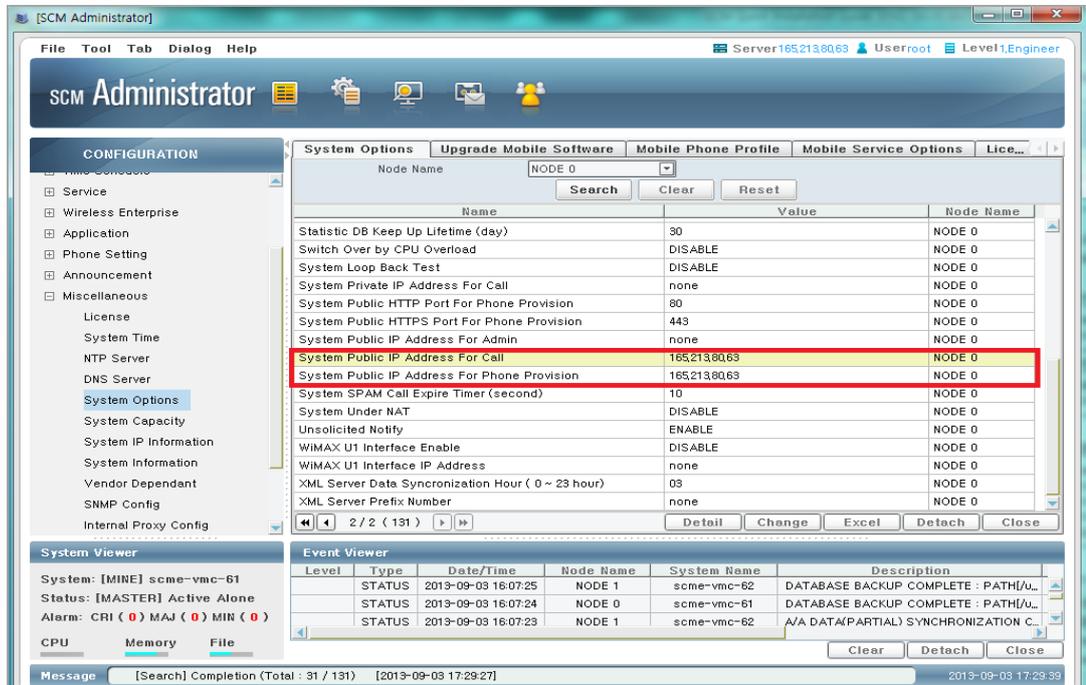
## Creating User

Create a user on the phone type set to Samsung-Mobile-Phone and set Use mVoIP option to Enable.



## Network Configuration

### Specify Public IP for mVoIP



## 16 TLS and sRTP configuration

### 16.1 SCM

#### TLS configuration

##### Configure TLS Version

SCM supports TLS V1.0 and V1.2. You can select one of them.

[CONFIGURATION → Miscellaneous → System Option → SIP TLS Version]

##### Configure TLS Use

- You can enable TLS of a Single Phone User by setting Protocol to TLS in the [CONFIGURATION > User > Single Phone User] menu.
- You can enable TLS of a Multi-Extension Phone by setting Protocol to TLS in the [CONFIGURATION > User > Multi-Extension Phone] menu.
- You can enable TLS for endpoints by setting Protocol to TLS in the [CONFIGURATION > Trunk Routing > Route] menu.

##### TLS Certification

You can be issued from Certificate Authority for SCM, phone, gateway and the issued Certifications should be located specific directory.

- 1) Directory Path  
/DI/BASE/data/COM
- 2) SCM Certification Type and name format
  - ① root\_caCert.pem: Root CA certification
  - ② caCert.pem: CA certification
  - ③ myCert.pem: PBX certification
  - ④ myPrvKey.pem: PBX private key

When install SCM, the basic three certifications-caCert.pem, myCert.pem, myPrvKey.pem are built-in.

If you get from external Certificate Authority, you must change name to above type.

If the private key is encrypted, you enable the option in following menu

[CONFIGURATION → Miscellaneous → System Option → SIP TLS Private Key Decryption → Enable].

To apply new certification, it is necessary to restart SCM.

### 3) Phone Certification

Phone can download own certification from SCM, if the certification is located in SCM specific directory and Use TLS Certification options is enabled.

**[CONFIGURATION > Phone Setting > SIP Options > Use TLS Certification > Enable]**

#### **Phone Certification Directory Path**

/tftpboot/sec\_cert

#### **Phone Certification Type and name format**

- {MAC}.der.pem: Phone certification
- {MAC}.key.pem: Phone private key
- example: phone's MAC is 00214c9c8eef  
certification: 00214c9c8eef.der.pem  
private key: 00214c9c8eef.key.pem

## **SRTP**

The media encryption feature provides encryption for the voice data exchanged between the caller and the called party for calls established with signaling encryption.

SCM supports media encryption for calls with phones, SCM's built-in conference system, SCM's built-in voice mail system, endpoints, and SCM's built-in MOH system.

SCM supports AES and ARIA as media encryption algorithms.

You can enable media encryption for a Single Phone User by setting 'Media' in the **[CONFIGURATION > User > Single Phone User]** menu. You can enable media encryption for a Multi-Extension Phone by setting 'Media' in the **[CONFIGURATION > User > Multi-Extension Phone]** menu.

- RTP: No media encryption.
- sRTP (AES/ARIA128): Encrypts media into the ARIA128 or AES protocol, and uses AES first.
- sRTP (ARIA128/AES): Encrypts media into the ARIA128 or AES protocol, and uses ARIA128 first.
- sRTP (AES/ARIA192): Encrypts media into the ARIA192 or AES protocol, and uses AES first.
- sRTP (ARIA192/AES): Encrypts media into the ARIA192 or AES protocol, and uses ARIA192 first.
- sRTP (AES): Encrypts media into the AES protocol.
- sRTP (ARIA128): Encrypts media into the ARIA128 protocol.
- sRTP (ARIA192): Encrypts media into the ARIA192 protocol.

Media encryption is not used for calls between phones or endpoints enabled with encryption and phones or endpoints not enabled with encryption.

## 16.2 OfficeServ7400

### TLS configuration

#### Configure TLS Version

SCM supports TLS V1.0 and V1.2. You can select one of them.

**[DM 5.2.12 SIP Stack/Ext/Trunk Options → SIP Trunk Configuration → TLS Version]** select V1.0 or V1.2

#### Configure TLS Use

**[DM 5.2.13 SIP Carrier Options → SIP Signal Type]** set to TLS

#### TLS Certification

You can be issued from Certificate Authority for SCM, phone, gateway and the issued Certifications should be located specific directory.

- 1) Directory Path  
**Located on SD card**  
**/certification/sip**
  
- 2) OS7400 certificate type and name format
  - ① root\_caCert.pem: Root CA certification
  - ② caCert.pem: CA certification
  - ③ myCert.pem: PBX certification
  - ④ myPrvKey.pem: PBX private key

OS7400 package involves the default certificates (rootCaCert.pem, caCert.pem, myCert.pem, myPrvKey.pem)

If you get from external Certificate Authority, you must change name to above type.

After changing the certificate, **[DM 5.2.13 SIP Server Enable]** Disable → Enable should be done.

### S RTP

OS7400 supports AES and ARIA of SRTP encryption algorithm. You can set for this in the DM.

**[DM 2.1.5 System Options → sRTP Algorithm]** select encryption algorithm

**[DM 5.2.16 MGI Options → USE sRTP]** set to Enable

## 16.3 iBG

### TLS configuration

You need to connect to Ubigate iBG via Console with terminal program. First of all, You must shutdown the voip gateway.

```
iBG# configure terminal  
iBG/configure/voip-gateway# shutdown
```

If you have to use certification for TLS, you need to upload certification file to CF card of iBG. After that, set the certification file.

In case of RootCA+CA+EndSystem

```
iBG# configure terminal  
iBG/configure# voice service sip tls-option load-3cert r c e k
```

In case of RootCA+EndSystem

```
iBG# configure terminal  
iBG/configure# voice service sip tls-option load-cert r e k
```

Change the transport option for TLS, and restart voip gateway.

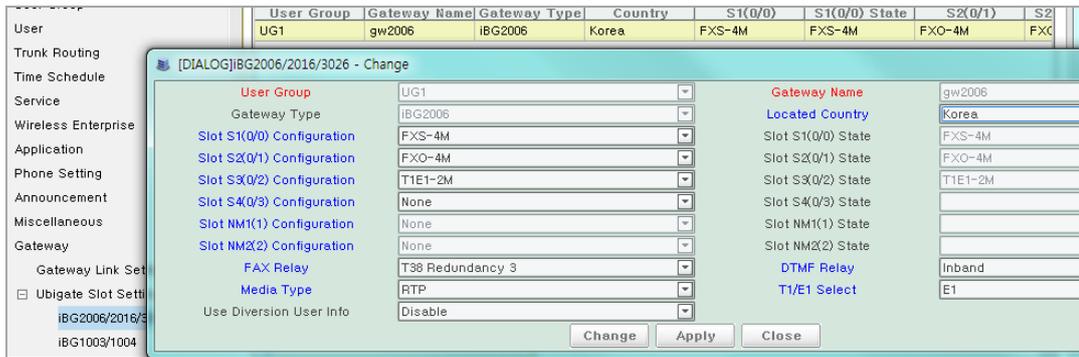
```
iBG# configure terminal  
iBG/configure# voice service sip transport tls  
iBG/configure# voip-gateway  
iBG/configure/voip-gateway# shutdown  
iBG/configure/voip-gateway# no shutdown
```

### SRTP configuration

Select the media type for SRTP configuration. This configuration will affect system globally. It is not possible to configure media type for individual FXS user or trunk.

**[Gateway > Ubigate Slot Setting > iBG2006/2016/3026]** or

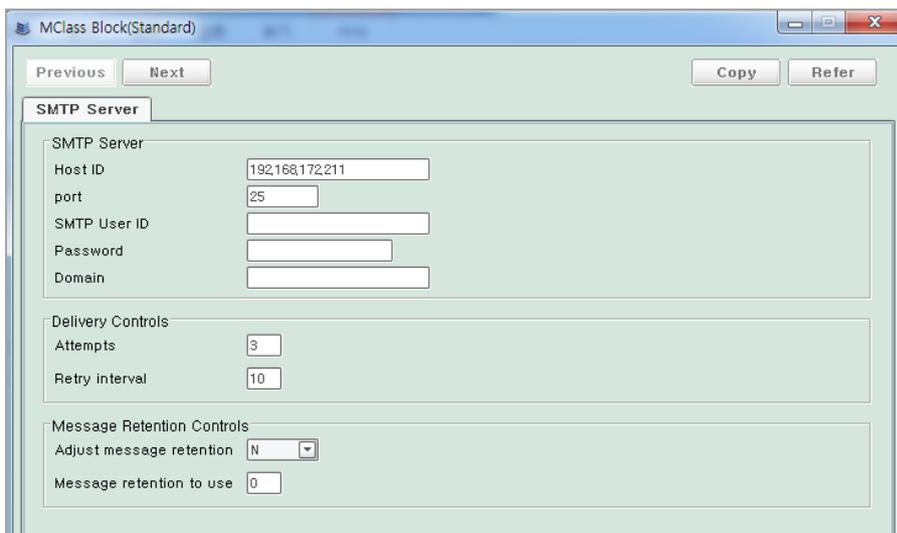
**[Gateway > Ubigate Slot Setting > iBG1003/1004]**



# 17 Interworking mail server

## E-mail Delivery and Notification for VM

SCM can send e-mail for void message. You can configure your email server in the [VM/AA > Open Block Table > Mailbox Class > SMTP Server]



Item	Description
Host ID	Enter the IP address of the Host Mail Server used by the users assigned this MClass.
PORT	The default (recommended) port to use is: 25. Most Mail Servers look at port 25 for receiving and sending Mail
SMTP USER ID (OPTIONAL)	This is the User ID the SCME VM will use to log on to the Mail Server and Identify itself as a Client associated with sending Mail (Mail Servers that are on a local LAN and that do not have Public IP addresses often do not require authentication)
PASSWORD (OPTIONAL)	This is the password associated with the SCME VM's User ID for logging into the Mail Server verifying it is the Client it said it was. (Mail Servers that are on a local LAN and that do not have Public IP addresses often do not require authentication.)
DOMAIN (OPTIONAL)	The Domain is used as part of the authentication process between the SCME and the Mail server. Based on the Local Domain Name and Domain ID the mail server can validate that it is accepting mail from this Client. (Mail Servers that are on a local LAN and that do not have Public IP addresses often do not require authentication.)
ATTEMPTS	How many times to do you want the SCME VM to attempt to deliver the E-Mail Message if it fails? The Default value is: 3.
RETRY INTERVAL	This is how long the SCME VM will wait between failure attempts before trying to deliver the e-mail message again

Item	Description
ADJUST MESSAGE RETENTION	'N' is the default setting. This means SCME VM will leave the original Voice Message as New. The Subscriber can then go in and Delete or Save the Voice Message via the telephone interface at any time up to the number of days specified in the Message Retention timer set on page one of the MClass. 'Y' means the SCME VM will follow the 'Message Retention to use:' value set below in place of the Message Retention set on page one
MESSAGE RETENTION TO USE	Sets the number of days to retain the Voice Message as New after it sends it to the Mail server. A value of '0' means delete the original voice message immediately after it is packed up and sent to the Mail Server. 'Adjust Message Retention:' must be set to 'Y' for this parameter to take effect. The selected range is from 0 to 999

### E-mail Notification Setup

SCM can send e-mail to notify Alarm/Fault/Status information or making conference reservation. You can configure your email server settings in the **[PERFORMANCE > Fault > E-mail Notification Setup]** menu.

Item	Description
SMTP Server: Host ID	IP address or host name of the e-mail server.
SMTP Server: Port	Port number of the e-mail server - Support SMTP or TLS (Start TLS) only - SMTP usually 25, TLS usually 587 * It can be different from e-mail server.
SMTP Server: Domain	Domain name of the e-mail server (optional)
Auth Login: User ID, Password	User ID and password required for user authentication by the e-mail server - In case Basic Authentication is set to ON in e-mail server, User ID and Password should be set. - In case Basic Authentication is set to OFF, User ID and Password should not be set. (For SCM Administrator not to try authentication)
Address: From	Sender's e-mail address
Address: To	Recipient's e-mail address * It's specified for notifying alarm/fault/status. * You can configure recipient's e-mail address for notifying conference reservation in <b>[Configuration &gt; User &gt; User Profile]</b> .

### Notifying Alarm/Fault/Status message

The alarm, fault and status information generated in SCM can be notified to the administrator by e-mail. You can set **[E-mail Flag]** to Enable in **[Performance > Fault > Setting > Setting-Alarm, Setting-Fault, Setting-Status]** by item.

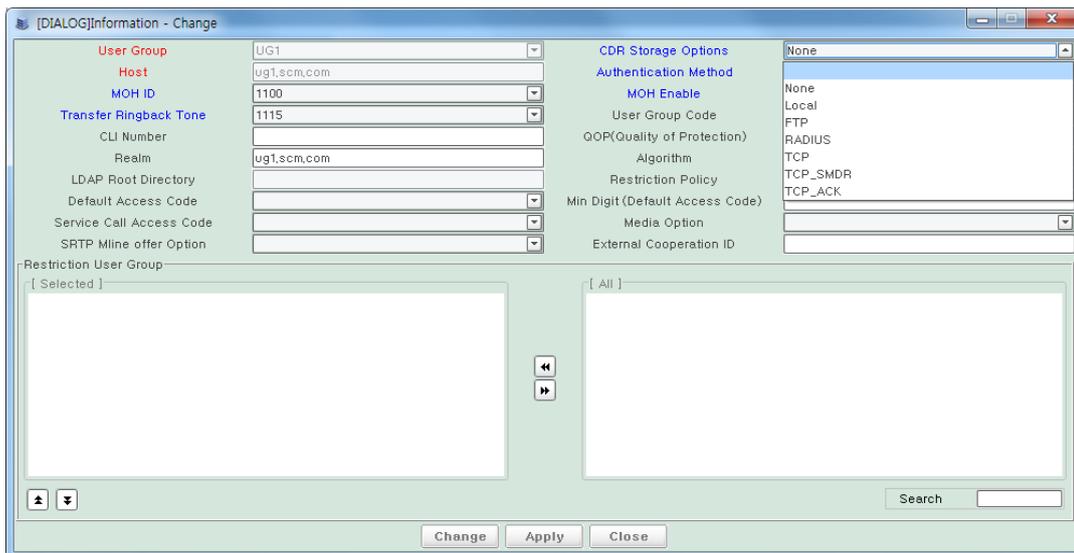
### Notifying Meet Me Conference Reservation

The invitation letters for Meet Me Conference Reservation can be sent to the members. You can check [**Send Invitation letters**] in [**Conference > Conference Management > Meet Me Reservation**].

# 18 Configuration of CDRs

## 18.1 Configuration of Storage option

The SCM creates and stores CDRs according to the specified configuration at CDR Storage Option in [CONFIGURATION > User Group > Change User Group > Information]. There are 7 CDR Storage options as below.



### NONE

The CDRs are not created.

### Local

The CDRs are stored at the local disk of the SCM.

For detailed configuration for 'Local', configure the following item.

[MANAGEMENT > CDR Storage Options > Local]

Name	Description
CDR Local Backup Interval (hour)	When configuring CDR files to be backed up in the hard disk, specify the backup interval. All CDR files generated are moved to the backup directory and the files in the local directory are deleted at this interval. Only the CDR files not saved in the backup directory will be left in the local directory.
CDR Local Backup Lifetime (day)	When configuring CDR files to be backed up in the hard disk, specify the number of days for which the backed up CDR files will be kept in the hard disk. At midnight everyday, the system automatically deletes any backed up CDR files which have passed the specified date.
CDR Local Backup Used	Specify whether to back up the generated CDR files in the hard

Name	Description
	disk. If enabled, the files are backed up in the /DI/CM/data/cdr/local/Backup directory.
CDR Local create Interval (min)	Specify the interval in minutes at which the CDR data files will be generated. New CDR files are generated at this interval. No CDR file will be generated if there is no CDR information for this period. The CDR files generated are saved in the /DI/CM/data/cdr/local directory.

## FTP

The CDRs are sent to the CDR server via FTP.

For detailed configuration for 'FTP', configure the following item

**[MANAGEMENT > CDR Storage Options > FTP Send]**

Attribute	Description
CDR FTP Backup Lifetime (day)	When interoperating with the accounting system over FTP, the CDR files generated can be backed up in SCM even after they have been transferred by FTP. Specify the number of days for which the backed up CDR files will be kept. At midnight everyday, the system automatically deletes any backed up CDR files which have passed the specified date.
CDR FTP Backup Used	When interoperating with the accounting system over FTP, the CDR files generated can be backed up in SCM even after they have been transferred by FTP. If enabled, the CDR files are backed up in the /scm_data/cdr/ftp/Backup directory in SCM.
CDR FTP Create Interval (min)	When interoperating with the accounting system over FTP, specify the interval at which the CDR files are generated. New CDR files are generated at this interval. No CDR file will be generated if there is no CDR information for this period. The CDR files generated are saved in the /DI/CM/data/cdr/ftp directory.
CDR FTP Directory	Specify the name of the directory in the FTP server where the files will be saved when transferring CDR files over FTP.
CDR FTP IP	Specify the IP address of the FTP server when transferring CDR files over FTP.
CDR FTP Login Name	Specify the login name of the FTP server when transferring CDR files over FTP.
CDR FTP Password	Specify the password of the FTP server when transferring CDR files over FTP.
CDR FTP Port	Specify the port number of the FTP server when transferring CDR files over FTP.
CDR FTP Secure	Specify whether to use Secure-FTP when transferring CDR files over FTP.

Attribute	Description
CDR FTP Transfer Interval (min)	When interoperating with the accounting system over FTP, specify the interval (in minutes) at which the CDR files are transferred. All CDR files generated are transferred over FTP and the successfully transferred files are moved to the backup directory at this interval. Only the CDR files not transferred over FTP will be left in the local directory.

## RADIUS

The CDRs are sent to the CDR server via RADIUS protocol.

For detailed configuration for 'RADIUS', configure the following item

**[MANAGEMENT > CDR Storage Options > RADIUS Send]**

Attribute	Description
CDR RADIUS Backup Lifetime (day)	When interoperating with the accounting system over RADIUS and backing up CDR files in SCM, the CDR files generated can be backed up in SCM even after they have been transferred to the RADIUS server. Specify the number of days for which the backed up CDR files will be kept. At midnight everyday, the system automatically deletes any backed up CDR files which have passed the specified date.
CDR RADIUS Backup Used	When interoperating with the accounting system over RADIUS, the CDR files can be backed up in SCM even after they have been transferred to the RADIUS server. If enabled, the CDR files are backed up in the /DI/CM/data/cdr/radius/Backup directory in SCM.
CDR RADIUS Create Interval (hour)	When interoperating with the accounting system over RADIUS and backing up CDR files in SCM, specify the interval (in minutes) at which the CDR files to be backed up are generated. The CDR files generated are moved to the backup directory, the files in the local directory are deleted, and CDR files with new names are generated at this interval. No CDR file will be generated if there is no CDR information for this period. The CDR files generated are saved in the /DI/CM/data/cdr/radius directory. Only the CDR files not saved in the backup directory will be left in this directory.
RADIUS Account Server IP	Specify the IP address of the RADIUS server when interoperating with the accounting system over RADIUS.
RADIUS Account Server Port	Specify the port number of the RADIUS server when interoperating with the accounting system over RADIUS.
RADIUS Account Used	Specify whether the CDR data will be sent to the RADIUS server when interoperating with the accounting system over RADIUS.

## CDR TCP

The CDRs are sent to the CDR server via TCP protocol.

For detailed configuration for 'TCP', configure the following item [MANAGEMENT > CDR Storage Options > TCP Send]

Attribute	Description
CDR TCP Backup Lifetime (day)	When interoperating with the accounting system over TCP and backing up CDR files in SCM, the number of days for which the backed up CDR files will be kept in SCM. At midnight everyday, the system automatically deletes any backed up CDR files which have passed the specified date.
CDR TCP Create Interval (min)	When interoperating with the accounting system over TCP and backing up CDR files in SCM, specify the interval (in minutes) at which the CDR files to be backed up are generated. New CDR files are generated at this interval. No CDR file will be generated if there is no CDR information for this period. The CDR files generated are saved in the /DI/CM/data/cdr/tcp directory.
CDR TCP Link1 IP	Specify the IP address of the first of the four TCP servers, to which the CDR data can be transferred when interoperating with the accounting system over TCP.
CDR TCP Link1 Used	Specify whether to transfer the CDR data to the first of the four TCP servers, to which the CDR data can be transferred when interoperating with the accounting system over TCP.
CDR TCP Link2 IP	Specify the IP address of the second of the four TCP servers, to which the CDR data can be transferred when interoperating with the accounting system over TCP.
CDR TCP Link2 Used	Specify whether to transfer the CDR data to the second of the four TCP servers, to which the CDR data can be transferred when interoperating with the accounting system over TCP.
CDR TCP Link3 IP	Specify whether to transfer the CDR data to the third of the four TCP servers, to which the CDR data can be transferred when interoperating with the accounting system over TCP.
CDR TCP Link3 Used	Specify whether to transfer the CDR data to the third of the four TCP servers, to which the CDR data can be transferred when interoperating with the accounting system over TCP.
CDR TCP Link4 IP	Specify the IP address of the fourth of the four TCP servers, to which the CDR data can be transferred when interoperating with the accounting system over TCP.
CDR TCP Link4 Used	Specify whether to transfer the CDR data to the fourth of the four TCP servers, to which the CDR data can be transferred when interoperating with the accounting system over TCP.

## TCP\_SMDR

Basically same as TCP options but the data is the format of SMDR.

For detailed configuration for 'TCP\_SMDR', configure the following item

**[MANAGEMENT > CDR Storage Options > Old SMDR Send]**

Attribute	Description
SMDR TCP Backup Lifetime (day)	When interoperating with the accounting system over TCP, send SMDR data and backing up CDR files in SCM, the number of days for which the backed up CDR files will be kept in SCM. At midnight everyday, the system automatically deletes any backed up CDR files which have passed the specified date.
SMDR TCP Create Interval (min)	When interoperating with the accounting system over TCP, send SMDR data and backing up CDR files in SCM, specify the interval (in minutes) at which the CDR files to be backed up are generated. New CDR files are generated at this interval. No CDR file will be generated if there is no CDR information for this period. The CDR files generated are saved in the /DI/CM/data/cdr/tcpSMDR directory.
SMDR TCP Link1 IP	Specify the IP address of the first of the four TCP servers, to which the SMDR data can be transferred when interoperating with the accounting system over TCP.
SMDR TCP Link1 Used	Specify whether to transfer the SMDR data to the first of the four TCP servers, to which the SMDR data can be transferred when interoperating with the accounting system over TCP.
SMDR TCP Link2 IP	Specify the IP address of the second of the four TCP servers, to which the SMDR data can be transferred when interoperating with the accounting system over TCP.
SMDR TCP Link2 Used	Specify whether to transfer the SMDR data to the second of the four TCP servers, to which the SMDR data can be transferred when interoperating with the accounting system over TCP.
SMDR TCP Link3 IP	Specify the IP address of the third of the four TCP servers, to which the SMDR data can be transferred when interoperating with the accounting system over TCP.
SMDR TCP Link3 Used	Specify whether to transfer the SMDR data to the third of the four TCP servers, to which the SMDR data can be transferred when interoperating with the accounting system over TCP.
SMDR TCP Link4 IP	Specify the IP address of the fourth of the four TCP servers, to which the SMDR data can be transferred when interoperating with the accounting system over TCP.
SMDR TCP Link4 Used	Specify whether to transfer the SMDR data to the fourth of the four TCP servers, to which the SMDR data can be transferred when interoperating with the accounting system over TCP.

## TCP\_Ack

Basically same as TCP options but the CDRs are sent to the TCP\_ACK CDR server via proprietary TCP protocol.

For detailed configuration for 'TCP\_ACK', configure the following item

[MANAGEMENT > CDR Storage Options > TCP ACK Send]

Name	Description
CDR TCP Ack Link1 IP	Specify the IP address of the first of the four TCP ACK servers, to which the CDR data can be transferred when interoperating with the accounting system over TCP ACK.
CDR TCP Ack Link1 Used	Specify whether to transfer the CDR data to the first of the four TCP ACK servers, to which the CDR data can be transferred when interoperating with the accounting system over TCP ACK.
CDR TCP Ack Link2 IP	Specify the IP address of the second of the four TCP ACK servers, to which the CDR data can be transferred when interoperating with the accounting system over TCP ACK.
CDR TCP Ack Link2 Used	Specify whether to transfer the CDR data to the second of the four TCP ACK servers, to which the CDR data can be transferred when interoperating with the accounting system over TCP ACK.
CDR TCP Ack Link3 IP	Specify the IP address of the third of the four TCP ACK servers, to which the CDR data can be transferred when interoperating with the accounting system over TCP ACK.
CDR TCP Ack Link3 Used	Specify whether to transfer the CDR data to the third of the four TCP ACK servers, to which the CDR data can be transferred when interoperating with the accounting system over TCP ACK.
CDR TCP Ack Link4 IP	Specify the IP address of the fourth of the four TCP ACK servers, to which the CDR data can be transferred when interoperating with the accounting system over TCP ACK.
CDR TCP Ack Link4 Used	Specify whether to transfer the CDR data to the fourth of the four TCP ACK servers, to which the CDR data can be transferred when interoperating with the accounting system over TCP ACK.

## Set Billing Output

This is a function that the CDR data is created by call types. To use this function, you should set the item whose 'use billing output' is 'ENABLE' in the **[MANAGEMENT > CDR Storage Options > CDR Option]** menu. And you should set items in the **[MANAGEMENT > CDR Storage Options > Set Billing Output]** menu.

Name	Description
App Server Call Billing Output [App Server]	The CDR is created if Calling type (4) is application and Called Type is application (4)
App Server Call Billing Output [Service]	The CDR is created if Calling Type (4) is Application and Called Type is Service (2)
App Server Call Billing Output [Subscriber]	The CDR is created if Calling Type (4) is Application and Called Type is Subscriber (1)
All Call Billing Ouput Set	The CDR is created about all calls
Incoming Call Billing Output[Normal]	The CDR is created if the trunk is a normal type and call is a outgoing call
Incoming Call Billing Output[TIE]	The CDR is created if the trunk is a TIE type and is call is a outgoing call
Outgoing Call Billing Output[Normal]	The CDR is created if the trunk is a normal type and call is a incoming call
Outgoing Call Billing Output[TIE]	The CDR is created if the trunk is a TIE type and call is a incoming call
Service Call Billing Output[App Server]	The CDR is created if Calling Type is Service (2) and Called Type is application (4)
Service Call Billing Output[Service]	The CDR is created if Calling Type is Service (2) and Called Type is Service (2)
Service Call Billing Output[Subscriber]	The CDR is created if Calling Type is Service (2) and Called Type is Subscriber (1)
Subscriber Call Billing Output[App Server]	The CDR is created if Calling Type is Service (1) and Called Type is application (4)
Subscriber Call Billing Output[Service]	The CDR is created if Calling Type is Service (1) and Called Type is Service (2)
Subscriber Call Billing Output[Subscriber]	The CDR is created if Calling Type is Subscriber (1) and Called Type is Subscriber (1)

## Length of Bill Delete

This is a function that deletes access code of trunk in the 'connect number' of CDR. To use this function, you should set 'Length of Bill Delete' in the **[Configuration > Trunk Routing > Route]** menu

## 19 OfficeServ phone upgrade

How to Upgrade the OfficeServ PKG phone to SCME PKG can be divided into upgrade from SCME System and upgrade from the menu on the terminal itself.

### Upgrade method from the terminal menu.

- Execute the HTTP Server or TFTP Server on the PC.
- Enter the IP address of the PC and Upgrade Type from the S/W Upgrade menu of the administrator menu of the terminal and press OK button, then Upgrade will be progressed.

### Upgrade method from SCME System

Uploading SCME PKG of terminal to the SCME system using the SCME Admin .



- PNP Mode Upgrade  
After terminal is completed booting, upgrade to SCME PKG by referring to the SCME IP information received from the DHCP Server.
- Static IP/DHCP Mode Upgrade  
Enter the necessary information of the terminal IP, Server IP (SCME IP), User ID, Password, through easy installation menu of the terminal, and then reboot the terminal. It will proceed to upgrade SCME PKG from SCME system.

## 20 Interworking application (SIP, CSTA)

### CSTA License

CSTA license defines the capability to use CSTA application. Operators should input CSTA licenses in the [SCM Administrator/CONFIGURATION/Miscellaneous/License] menu to integrate with CSTA applications such as Samsung Operator, Embedded ACD Agent, Communicator, ACD Server, SC Plus and other CSTA Applications.

When an external license key is issued, Samsung Operator, Embedded ACD Agent and Communicator can be set the number of CSTA application channels for the each corresponding entries. But this cannot exceed the number of Total CSTA applications entry. ACD Server and SC Plus do not need licenses. Because SCM provides 10 license for ACD server and SC Plus. Every CSTA applications without previously mentioned uses the channels specified in [Other CSTA Applications].

The number specified in [Total CSTA Applications] entry means total available number that can be assigned to CSTA applications. It is including 10 licenses that SCM basically provides for ACD server and SC Plus.

Although only ACD server or SC Plus that does not need to input licenses is used, operator should input External Application License.

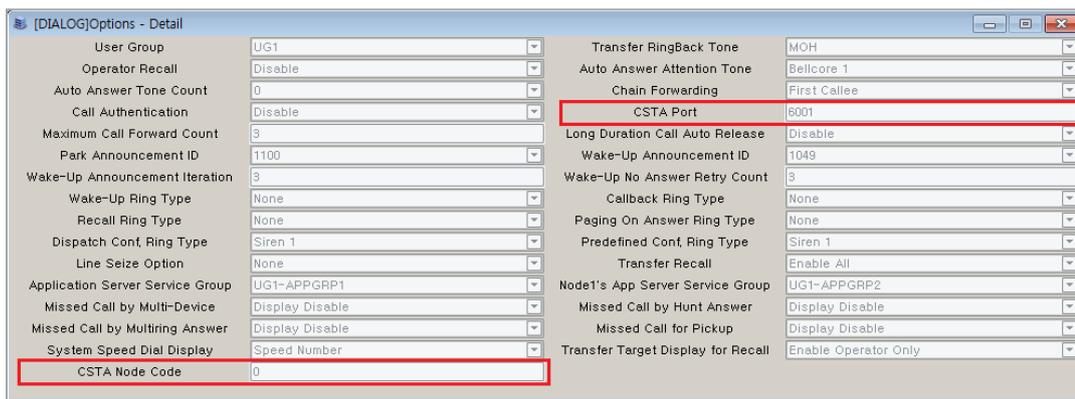
[DIALOG]License - Detail	
License Key Type	SCM Express - External Application
MAC Address	
Samsung SIP Phones	
Samsung Mobile Phones	
3rd Party SIP Phones	
AA Availability(Master/Slave)	
Meet-Me Conference Channels	
<b>Total CSTA Applications</b>	<b>50</b>
<b>Embedded ACD Agent Links</b>	<b>30</b>
<b>Other CSTA Applications</b>	<b>8</b>
FMS Phones	
License Key	Time: 2011/11/07 14:18:22, Elapsed Time: 8205(hr)
License Status	Using time expired (30days)
Samsung Soft Phones	
Samsung PC Attendants	
Analog Phones(Gateway)	
High Availability(Active/Standby)	
UMS Channels	
<b>Samsung Operators</b>	<b>2</b>
<b>Communicators/Desktop</b>	<b>10</b>
SIP Application Channels	100
MVS Phones	

### CSTA Configuration

As a client, CSTA application makes TCP connection to SCM with SCM IP address and CSTA port.

Each User Group has separate connection with a different CSTA port, which is determined according to the User Group. Therefore, ACD Group/Agent/IVR is controlled depending on the User Group. CSTA port for each user group is shown in the [SCM Administrator/CONFIGURATION/User Group/Change User Group/Options] menu.

When several SCM is installed in a site, unique CSTA Call Id is available by setting CSTA Node Code in the [SCM Administrator/CONFIGURATION/User Group/Change User Group/Options] menu. Each SCM should have different CSTA Node Code. If just one SCM is installed in a site, CSTA Node Code has no meaning.



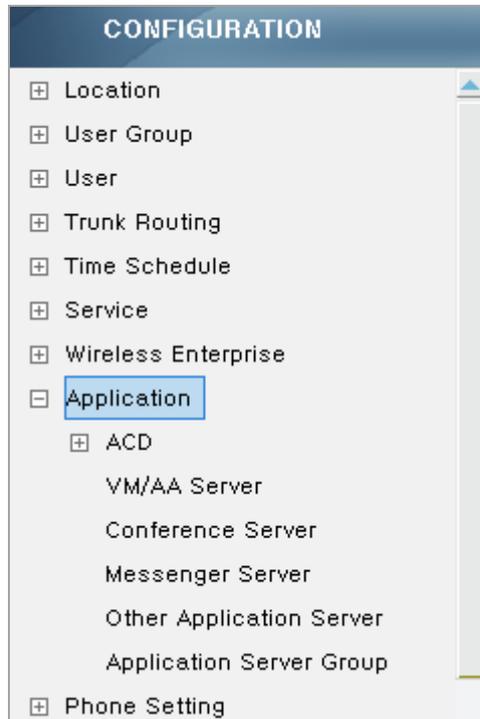
### SIP Application License

SCM License scheme defines SIP application channel capabilities. Operators can insert the external license key issued by a license server using **[SCM Administrator/ CONFIGURATION/Miscellaneous/License]** menu. In order to integrate with application servers such as Voice Mail, Recording, Auto Attendant, Conference and External Ringback Tone server, the **[SIP Application Channels]** of External License should be assigned. The **[SIP Application Channels]** defines SIP application channel capabilities. Each application needs to set SIP channels in each application menu. The number of each application license cannot exceed the total number of SIP application channel capacity. Total SIP application channels are be shown as below.



## Creating Application Server

- 1) Create Application Server  
**[CONFIGURATION > Application]**



- 2) Create Application Server Group  
 When creating Server Group, Each Service must be set by Application Server.  
**[CONFIGURATION > Application > Application Server Group]**



3) Set Application Server Group to User Group, Service Group, User  
**[CONFIGURATION > User Group > Change User > Options]**

[DIALOG] Options - Change

User Group	UG1	Transfer RingBack Tone	MOH
Operator Recall	Disable	Auto Answer Attention Tone	Bellcore 1
Auto Answer Tone Count	0	Chain Forwarding	First Callee
Call Authentication	Disable	CSTA Port	6001
Maximum Call Forward Count	3	Long Duration Call Auto Release	Disable
Park Announcement ID	1100	Wake-Up Announcement ID	1049
Wake-Up Announcement Iteration	3	Wake-Up No Answer Retry Count	3
Wake-Up Ring Type	None	Callback Ring Type	None
Recall Ring Type	None	Paging On Answer Ring Type	None
Dispatch Cont. Ring Type	Siren 1	Predefined Cont. Ring Type	Siren 1
Line Seize Option	None	Transfer Recall	Enable All
<b>Application Server Service Group</b>	<b>UG1-APPGRP1</b>	Node1 App Server Service Group	Node1 App Server Service Group
Missed Call by Multi-Device	Display Disable	Missed Call by Hunt Answer	Display Disable
Missed Call by Multiring Answer	Display Disable	Missed Call for Pickup	Display Disable
System Speed Dial Display	Speed Number	Transfer Target Display for Recall	Enable Operator Only
CSTA Node Code	0	No Ring for Multi-Device	Disable
Minimum Number Translation Length	10	Hunt Group Member Service	Disable
Smart Routing Service	Enable	Use Default Access Code Use List	No

Change Apply Close

**[CONFIGURATION > User Group > Service Group]**

[DIALOG] Service Group - Change

User Group	UG1	Name	UG1-SG1
Service Group Code		CLI Number	
Class of Service		Restriction Policy	
Dial Tone		Dial Plan	
<b>Application Server Service Group</b>	<b>UG1-APPGRP1</b>	Node1 App Server Service Group	Node1 App Server Service Group
Call Recording Method	Conference Recording	Auto Attendant Ring Plan Schedule	
CFUR Service Schedule			

Change Apply Close

**[CONFIGURATION > User > Single Phone User]**

[DIALOG] Single Phone User - Change

User Group	UG1	Service Group	UG1-SG1
Location	UG1-LOC1	Extension Number	2001
Application User ID	2001@ug1.scm.com	Extension Name	2001
Application Password	*****	PIN Number	****
Authentication User ID	2001	Phone Verification	None
Authentication Password	****	MAC Address	
IP Address	10.251.191.164	Private IP Address	10.251.191.164
Profile Login ID	UG12001	Phone Type	Samsung-Desktop-Phone
Profile Login Passcode	****	Language	Korean
Mobile Phone Number		Use Mobile Phone Number	None
Protocol	UDP	Media	RTP
TLS Connection	Reuse	Ping Ring Type	None
A-A Primary Node	NODE 0	A-A Dual Registration	Enable
VMS Extension Number		Make Mailbox	Yes
URI Type	SIP	DTMF	RFC2833
RFC2833 DTMF Payload	101	Time Zone	GMT +09:00 Asia/Seoul
Department		Position	
Send CLI Number		Service Group Local CLI Number	
Service Group Local Number		Restriction Policy	
Class of Service		Gateway Name	
Extension Lock	None	LDAP DN Number	
Account Code Use		Auto Answer by Click to Dial	Enable
Accept Login Override	Disable	External Ringback Tone Use	None
MOH Announcement ID		Display Option	Normal
Send CLI Name		Call Monitoring	Disable
Send Extension Number		Use Virtual Ringback	Disable
Caller Ring Type	None	Off Hook Alarm	
Check Registration Protocol	Disable	MOH SIP Media Mode	Send Only
<b>Application Server Service Group</b>	<b>UG1-APPGRP1</b>	CMS Monitoring	Disable

Change Apply Close



# ABBREVIATION

## A

AA	Auto Attendant
AAR	Automatic Alternative Routing
ACD	Automatic Call Distribution
AR	Alternative Route

## B

BHCA	Busy Hour Call Attempt
BLF	Busy Lamp Field

## C

CAC	Call Admission Control
CDR	Call Detailed Record
CLI	Calling Line Identification
CLIR	Calling Line Identification Restriction
COS	Class of Service
CPS	Call Per Second
CSTA	Computer Supported Telephony Application
CTI	Computer Telephony Interface

## D

DID	Direct Inward Dial
DISA	Direct Inward System Access
DN	Directory Number
DND	Do Not Disturb
DOD	Direct Outward Dial
DR	Direct Route
DTMF	Dual Tone Multi-Frequency

## F

FMC	Fixed Mobile Convergence
-----	--------------------------

**I**

ITSP	Internet Telephony Service Provider
IVR	Interactive Voice Response

**L**

LDAP	Lightweight Directory Access Protocol
------	---------------------------------------

**M**

MCS	Multimedia Conference System
MOH	Music On Hold
MWI	Message Waiting Indication

**N**

NMS	Network Management System
-----	---------------------------

**P**

PBX	Private Branch eXchange
PSTN	Public Switched Telephone Network

**R**

RADIUS	Remote Authentication Dial In User Service
RFC	Request For Comments
RTP	Realtime Transport Protocol

**S**

SBC	Session Border Controller
SCM	Samsung Communication Manager
SIP	Session Initiation Protocol
SNMP	Simple Network Management Protocol

**T**

TLS	Transport Layer Security
-----	--------------------------

**U**

UMS	Unified Messaging System
-----	--------------------------

**V**

VMS	Voice Mailing System
VoIP	Voice over IP

# W

WE Wireless Enterprise

## **SCM**

# **Quick Installation Guide**

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